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THE ULTIMATE GUIDE TO ABLETON LIVE

A COMPUTER MUSIC SPECIAL EDITION

ABLETON LIVE **THE ULTIMATE GUIDE**

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welcome

Over the last decade and a half, Ableton's amazing music production software, Live, has played a huge part in shaping the modern music technology landscape, both in the studio and on stage. Its streamlined workflow and innovative approach to audio timestretching genuinely changed the game when it first arrived in 2001, and over the years since, its design has influenced other DAW developers more profoundly than any of them would probably care to admit.

While Live is one of the most intuitive music applications around, it's also incredibly deep and powerful – which is where this hefty tome comes in, of course. *Ableton Live: The Ultimate Guide* is packed with walkthroughs, tips and technical advice, all put together to make your Live sessions (and, indeed, Sessions) more creative, productive and fun.

If you're totally new to the world of Live, our 39-page *Beginner's Guide* (p6) is the place to start, taking you through all the software's basic operations and functions, and priming you for the more advanced stuff to follow. The Live veteran, meanwhile, can work through the *Beginner's Guide* as a refresher course (I bet you'll rediscover at least a couple of features you'd forgotten about), or plough straight into the masterclasses, most of which also come complete with videos, bringing them to life on screen.

Whatever level you're coming at *Ableton Live: The Ultimate Guide* from, enjoy!

Ronan Macdonald, **Editor**

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THE ULTIMATE GUIDE TO ABLETON LIVE

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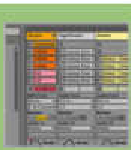

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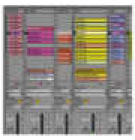

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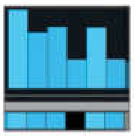

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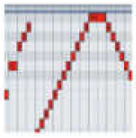
78 GET EXPERIMENTAL!



Sometimes, getting your crazy on is the only way to break out of that creative rut. Here, we present a range of conceptually adventurous ideas over the course of producing a complete track

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Turn lacklustre hi-hat lines into rhythmic and sonic gold with these ten detailed video tips – your drum loops will thank you for it!

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Live's awesome hybrid synth is capable of far more than just FM-style tones. We show you how to press it into service for kick drums, faux analogue bass, ambient pads and more

100 MASTER BUS FX

Cunning processing of the master output is the key to successful breakdowns, builds and DJ-style FX. Catch the bus right here!

107 ESSENTIAL TRANSITIONS

Moving seamlessly and musically from one section of a track to another can be a challenge – unless you've got our ten video tips to hand, that is

DOWNLOAD VIDEOS, FILES AND PLUGINS

To download the tutorial files and videos that accompany most of the tutorials in *Ableton Live: The Ultimate Guide*, as well as the CM Studio suite of software instruments and effects, visit the **cm** Vault. **vault.computermusic.co.uk**

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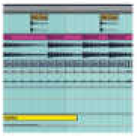

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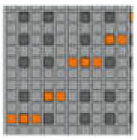

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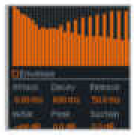

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DOWNLOAD

Get the tutorial files you need on your PC/Mac at **vault.computermusic.co.uk**

THE BEGINNER'S GUIDE TO ABLETON LIVE

New to Live? Prepare to experience the most powerful music production and performance software ever created!

Whether Ableton Live is your first experience of computer music-making or you've switched to Live from another DAW (Digital Audio Workstation), you've made a great choice – Live is an incredible piece of software that's used the world over by cutting-edge artists to record, produce and perform. Whatever it is that you want to accomplish with your music, Live is a fantastic tool, promoting creativity with its hassle-free workflow and inspirational range of features.

Because Live began its existence as a live performance application, its design is geared towards accessibility and transparency. However, those unfamiliar with the software can find its duality confusing. Live's innovative design is centred on two views: the grid-based Session View, which the software displays by default, and the more familiar Arrangement View, with its vertically arranged tracks. Users accustomed to other DAWs will feel much more at home when they discover that the Arrangement view can be switched to with the tap of a key or click of a button, though computer music novices will need a little more help to get their heads around some of the software's key concepts.

In this exhaustive guide we'll give you a thorough understanding of Ableton Live's core functionality, with walkthroughs giving straightforward, step-by step instruction on how to use practically all of its capabilities. We'll cover everything from making your first simple sequences and triggering sounds, to more advanced techniques such as FM synth programming and sidechain routing, and explain all the relevant music technology terminology along the way.

Once you've worked your way through the next 37 pages, and the easily digestible morsels of music-making goodness contained therein, you'll be more than ready to start performing, composing and producing with Ableton Live!

Let's get started...

Arrangement view

Session view

Live's Session and Arrangement views at a glance

01 SHOW/HIDE BROWSER
Toggles the Browser view

02 SEARCH
Enter text here to filter the contents of the Browser Content Pane

03 CATEGORIES
Filters down to items of a specific type in the Browser Content Pane

04 PLACES
Shows the contents of folders on your computer in the Browser Content Pane

05 BROWSER CONTENT PANE
Displays the effects, instruments, racks and sounds in the selected location

06 PREVIEW TAB
Preview settings and waveform display for audio files selected in the Browser Content Pane

07 GROOVE POOL
Contains all grooves used in the current project

08 INFO VIEW
Displays information on the GUI element currently under the mouse pointer

09 SHOW/HIDE INFO VIEW
Toggles the Info View on and off

10 DEVICE VIEW
Displays the currently selected track's device chain

11 AUDIO TRACK
A track for playing back audio clips. The red line is an automation envelope

12 MIDI TRACK
A track for playing back MIDI clips. The broken red line shows that the selected parameter isn't currently automated

13 BEAT TIME RULER
The bars/beats timeline.Drag on it to zoom in and out of the arrangement

14 LOCATOR
Click to start playback from this point. Can also be assigned to a key command or MIDI controller

15 LOCATOR CONTROLS
Add Locators and cycle through all of the Locators in the project

16 LOCK ENVELOPES
Prevents edits made to clips from affecting automation envelopes

17 TRACK TITLE BAR
Includes the Fades/Devices Chooser and Automation Control Chooser, which are used to select the parameter automation envelope displayed on the track

18 IN/OUT SECTION
Displays the track's input and output settings

19 MIXER SECTION
Contains Mute, Solo, level, Pan, Send and Record Arm settings for every track

20 TRACK DELAY
Set positive or negative playback time delay in samples or milliseconds

21 MASTER TRACK
Insert effects here to process the whole mix

22 RETURN TRACKS
Auxiliary channels for hosting send effects

23 SHOW/HIDE BUTTONS
Reveal and hide the In/Out Section, Return Tracks, Mixer and Track Delay panels

24 CLIP OVERVIEW/ZOOMING HOT SPOT
Zooms in and out of the currently selected clip

25 DEVICE VIEW SELECTOR
View the selected track's device chain, and drag to scroll around it

26 SHOW/HIDE DETAIL VIEW
Show and hide the Clip/Device view

27 SCRUB AREA
Click in this area to jump playback to that point

28 TRACK NAME
Click to select a track; double-click to open the Device view

29 ARRANGEMENT/SESSION VIEW SELECTOR
Toggle between the Arrangement and Session views with these two buttons

30 BACK TO ARRANGEMENT
When you're working in the Session view, this button reverts to the arrangement in the Arrangement view

31 CLIPS
Audio and MIDI clips are represented as blocks in the Session view

32 CLIP LAUNCH
This button starts playback of this clip

33 SCENES
A row of clips is called a Scene, and Scenes are triggered by these playback buttons

34 CLIP STOP
Clicking this button stops playback of this clip

35 STOP ALL CLIPS
Click this button to kill playback of all clips

36 CROSSFADE ASSIGN
Assign each track to the A or B crossfader channel

37 CROSSFADER
Fade Live's output between the A and B crossfader channels with this slider

38 SOLO/CUE
Toggle in and out of Cue mode for DJing purposes

39 PREVIEW/CUE VOLUME
Set the preview/cue channel volume level

40 CLIP VIEW
Properties and parameters of the currently selected audio or MIDI clip

The Control Bar close up

01 TAP TEMPO
Click here repeatedly and Live will set the project tempo to the timing of your tapping

02 TEMPO
The project tempo. Click to enter values directly

03 TEMPO NUDGE UP/DOWN
Briefly increase or decrease the project tempo to maintain sync with musicians or other sources using these two buttons

04 TIME SIGNATURE
Set the number (left numeral) and note value (right numeral) of beats in a bar

05 GLOBAL GROOVE AMOUNT
Scales the overall influence of grooves applied from the Groove Pool

06 METRONOME/COUNT-IN
Toggles the metronome on and off, and sets the length of the pre-record count-in

07 QUANTIZATION
Selects the global quantise resolution, which controls the timing of clip playback

08 FOLLOW
When this is active, the Arrangement view scrolls to follow the playback position

09 ARRANGEMENT POSITION
Displays the playback position in the Arrangement view. Drag to adjust playback position

10 TRANSPORT BUTTONS
Play, Stop and Record buttons

11 MIDI ARRANGEMENT OVERDUB
When active, new MIDI recordings will add to rather than replace existing MIDI clips

12 AUTOMATION ARM
With Live recording, when this is active, all parameter adjustments are captured as automation

13 RE-ENABLE AUTOMATION
Adjusting any automated parameter lights this button up; clicking it returns to the existing Arrangement or Session clip automation

14 LOOP START/PUNCH-IN
The start point of the loop or punch-in region

15 PUNCH-IN SWITCH
Disable Arrangement recording prior to the punch-in point when active

16 LOOP SWITCH
Toggle the Arrangement loop on/off

17 PUNCH-OUT SWITCH
Prevents recording into the Arrangement after the set point

18 LOOP/PUNCH REGION LENGTH
The length of the loop or punch-in region

19 DRAW MODE SWITCH
Toggles Draw Mode on and off

20 COMPUTER MIDI KEYBOARD
Enables your QWERTY keyboard to be used for note input when active

21 KEY MAP MODE
Toggles Key Map Mode on and off, for creating remote control assignments

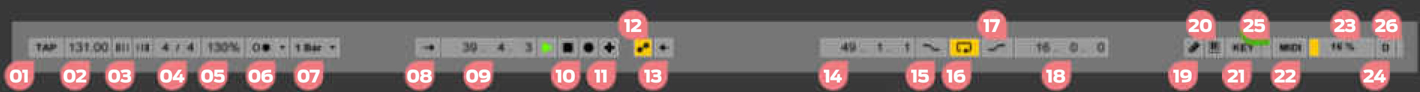
22 MIDI MAP MODE
Toggles MIDI Map Mode on and off, for creating remote control assignments

23 CPU LOAD METER
Keep track of your system usage and overhead here

24 HARD DISK OVERLOAD INDICATOR
Lights up if Live is struggling to load audio from your hard drive in time for accurate playback

25 KEY/MIDI IN/OUT INDICATORS
These light up when MIDI information relating to remote-control assignments is sent or received

26 MIDI TRACK IN/OUT INDICATORS
These light up when tracks are sending or receiving MIDI messages



> Step by step

1. Preparing Live for first-time use with the Preferences

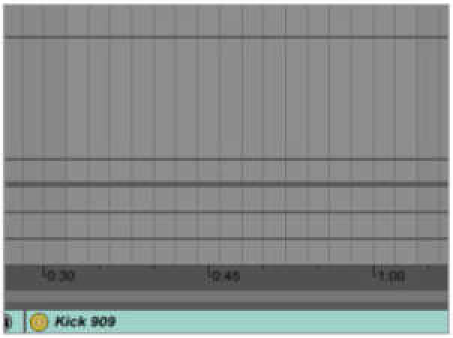
1 > The first time you run Live, you'll be presented with the **Improve Your Experience with Live** dialog. You can turn off automatic updates and the sending of usage data to Ableton here if you wish – changes can also be made to these settings in the **License/Maintenance** section of the **Preferences** menu at any time. Click the **OK** button to continue.

2 > The next thing to do is tailor the DAW to your liking with the **Preferences** menu – go to **Live»Preferences** in OS X or **Options»Preferences** in Windows. The **Look/Feel** page will be displayed by default. We suggest deactivating the **Auto-Hide Plug-In Windows** option to prevent Live from closing plugin windows when you select a new track.

3 > Click the **Audio** tab on the left-hand side of the window. Here, you can select an audio driver. If you're in Windows and working with your computer's built-in audio system, ASIO4ALL is a great universal driver that can be downloaded for free from **www.asio4all.com**. You can activate specific inputs and outputs on your audio interface with the **Input Config** and **Output Config** buttons.

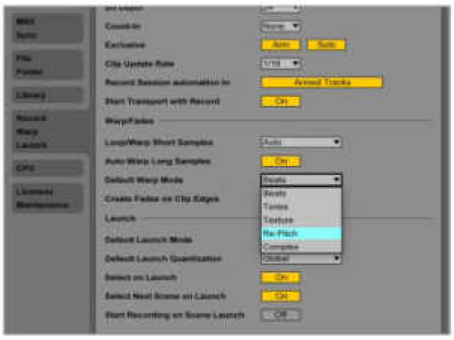
4 > Unless you're working with video, it's a good idea to set the **In/Out Sample Rate** to **44100**; this will ensure that the audio you record will be at the standard rate for music production. You can set your audio interface's buffer size here too – **512** or **1024** samples makes a good starting point. See *Size Matters*, below right, for a more involved method of determining optimal buffer size.

5 > In the **File/Folder** tab, you can set up your shared plugin folder or folders. If you're using the default plugin folders on a Mac, activate the **Use Audio Units Plug-In System Folders** and **Use VST Plug-In System Folders** buttons.



6 > If you're using a custom VST folder or you're on a Windows PC, click the **Use VST Plug-in Custom Folder** button. You'll be prompted to specify a location for the folder. Live will scan your hard drive for compatible plugins, with details displayed at the bottom of the interface.

7 > When you've finished adding plugin folders, click the **Record/Warp/ Launch** tab. We highly recommend deactivating **Create Fades on Clip Edges** in the **Warp/Fades** section – if you leave this on, Live will automatically add short fades to the start and end of all audio clips.



8 > It's also possible to alter how Live handles the warping of both short (eg, drum loops) and long (eg, a full song) audio clips, and change the **Default Warp Mode** from **Beats**. Leave these as they are for now, but don't forget about them, as you might want to customise them after become more accustomed to how Live operates. When you're done, close the **Preferences** window.

POWER TIP

>Size matters

Buffer size determines how much time your computer gets to generate and process audio: the larger it is, the more audio tracks, effects and virtual instrument can be crammed into a project before your CPU starts to overload and glitches occur. The downside is that latency increases with buffer size, and one of the ways latency manifests itself is in the lag between playing a note on your MIDI controller and hearing the resulting sound. Start with a buffer size small enough that you don't notice any latency, and turn it up when your CPU can't cope. You can monitor CPU usage via the CPU Load Meter at the top right-hand corner of Live's interface.

Sequencing your first beat with a MIDI track

If you're a relative newcomer to the world of computer music, you'll likely have noticed the acronym MIDI being bandied about a fair bit but perhaps not know what it means. MIDI stands for Musical Instrument Digital Interface, the most commonly used standard by which digital equipment communicates musical information. Don't worry if that sounds a little intimidating, because MIDI makes composing music in software a more intuitive experience than it would otherwise be. The way most software (including Live) handles MIDI data editing is based around the 'piano roll' or 'grid' model, which is very visual and much simpler for newbies to get to grips with than sheet music.

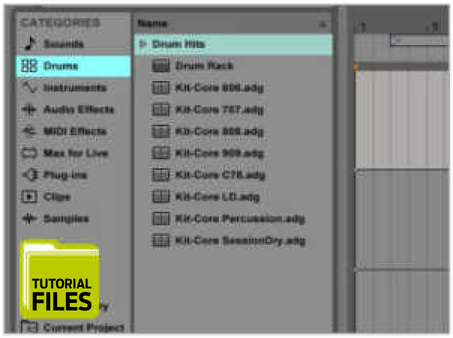
Both Live's Session and Arrangement views enable the user to create musical phrases known as MIDI clips, which can be arranged together to create a sequence. MIDI tracks can be used to trigger external hardware, or you can use plugin instruments 'in the box' – ie, virtual instruments hosted by your DAW. As well as third party plugins, Live can host a selection of built-in instruments which can be used to generate sounds – see *Instruments* on p33.

In this walkthrough, we'll look at how we can add an instrument to a MIDI track, use your computer's QWERTY keyboard to play it in real time, record your playing and edit the result with Live's MIDI Editor.

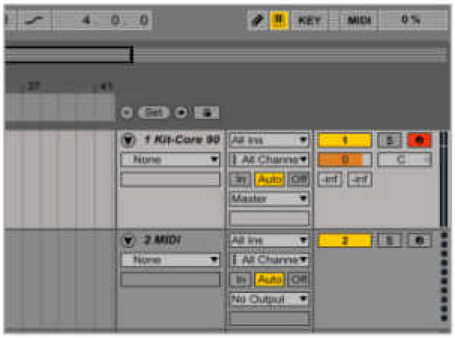
One of MIDI's many plus points is that it makes it easy to correct mistakes made when recording. For example, by moving notes to their nearest beat division, we can get a much tighter-sounding performance. Of course, perfect timing isn't always desirable, so we'll also show you how to disable Live's 'snap to grid' mode and add a bit of groove to your rhythms. For more on musical timing see *Tightening tracks with quantisation* on p17 and *The Groove Pool* on p27.

MIDI controllers come in all shapes and sizes, from keyboards and drum pads to mixer-style units

> Step by step 2. Making a beat on a MIDI track

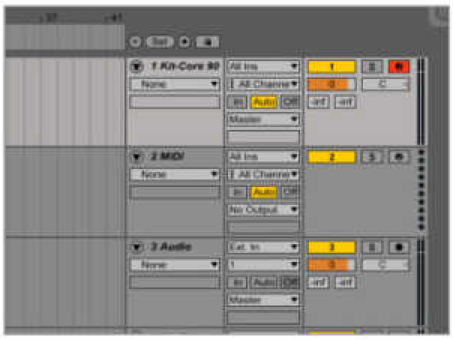


1 When you launch Live, you're presented with the Session view. Press **Tab** or click the **Arrangement View Selector** button, top right, to switch to the Arrangement view. In the **Categories** panel, click the **Drums** category to make a selection of kits available in the Browser Content Pane.

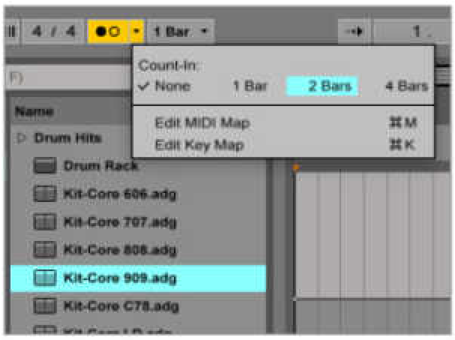


2 Drag **Kit-Core 909.adg** onto the first MIDI track, 1 MIDI. The **Kit-Core 909** Drum Rack preset will load, and you'll be able to play its sounds back using Live's Computer MIDI Keyboard function. Tap **A**, **S** and **D** on your computer's QWERTY keyboard. If you don't hear anything, check that the **Computer MIDI Keyboard** button is activated.

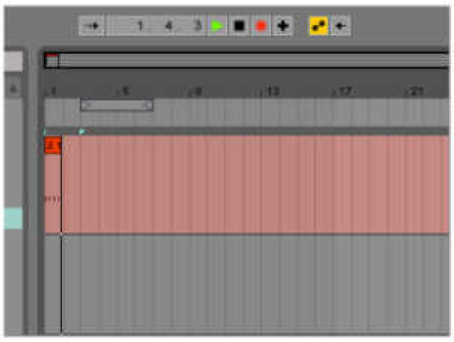
3 If it is and you still can't hear anything, make sure you're playing in the correct octave. Incoming MIDI notes are displayed by the Pad Overview on the left-hand side of the Drum Rack. The light-grey pads have sounds loaded and light up yellow-green when played. Adjust the octave range of the Computer MIDI Keyboard up and down with the **Z** and **X** keys.



4 If there's still no sound, check your audio driver settings, and that your speakers are turned on and their volume is turned up. Once you can hear the Drum Rack's output, you're ready to record a beat. Check that the **Arm Arrangement Recording** button on the MIDI track is red, which means it's active. If it's not, click it to activate it.



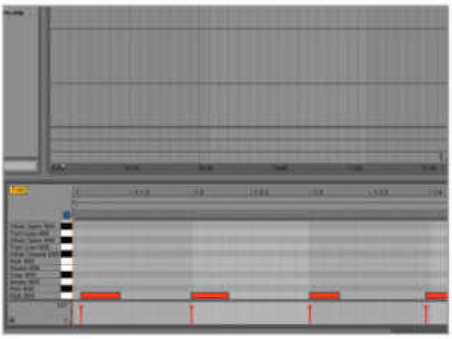
5 Click the **Metronome** button at the top left-hand corner of the interface to turn it on, then click the **Count-In** button to the right and select **2 Bars**. This gives us a two bar count-in. Now we're ready to record our first beat! Let's lay down a simple kick drum part. Set the Computer MIDI Keyboard's octave so that pressing **A** triggers the **Kick 909** pad.



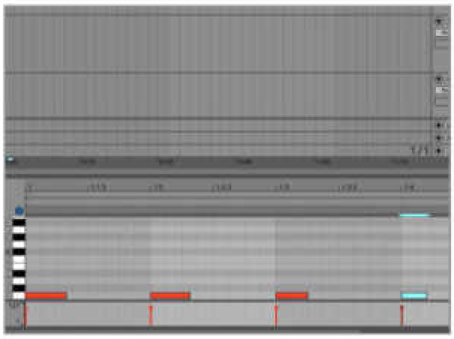
6 When you're all set, click the **Record** button. After the two-bar metronome count-in, play along on the beat with the metronome for a bar, then press the **Spacebar** to stop recording. A short MIDI clip will have been created. Double-click it to open the MIDI Editor in the Clip View.

> Step by step

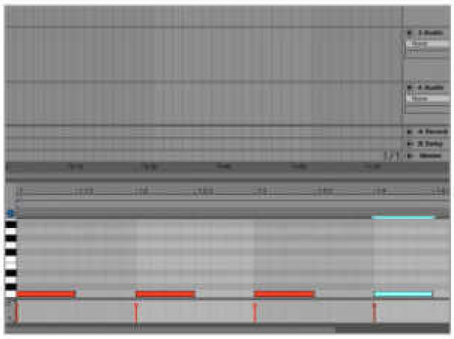
2. Making a beat on a MIDI track (continued)



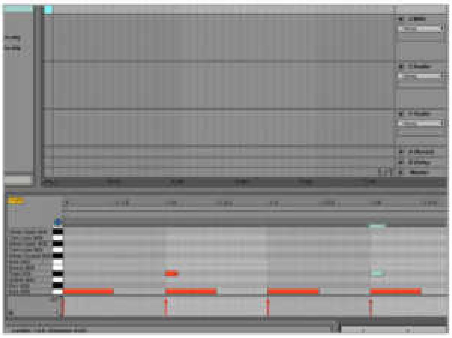
7 > If you can't see **Kick 909** in the list of drum sounds in the MIDI Editor, scroll the list by dragging it. **Kick 909** is at the bottom and our four beats have been recorded, although, unless you happen to be a robot, their timing won't be perfect.



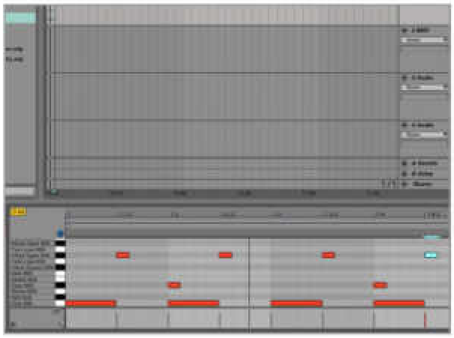
8 > Now let's fix the timing of the notes. We could do this automatically with some quantisation, but for the moment, let's do it manually. To adjust the timing of a note, simply drag it left or right. Drag the four kicks so that they line up with **1**, **1.2**, **1.3** and **1.4** on the timeline at the top of the MIDI Editor.



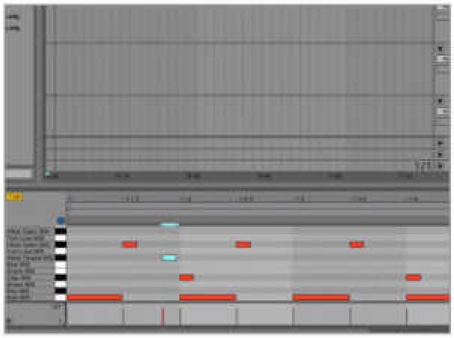
9 > You'll notice that the MIDI Editor's grid is 'sticky', making it easy to snap notes accurately to the beat. It's likely your notes will be of varying lengths, but we can make them uniform by dragging their right-hand edges to set their lengths. Make each note an eighth of a bar long, as shown above. (Audio: **Uniform kicks.wav**)



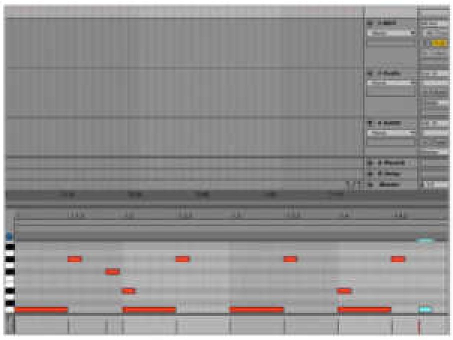
10 > Let's look at a couple of other ways to add more notes. Double-click the **Clap 909** row on beats **1.2** and **1.4**. If you make a mistake, double-click a note to delete it. To have a listen to what we've got so far, press the **Spacebar** or click the **Play** button. You'll notice our sequence just plays once. To loop it, drag over the bar with the MIDI clip on it on the MIDI track, and press **Ctrl/Cmd+L**. (Audio: **Claps.wav**)



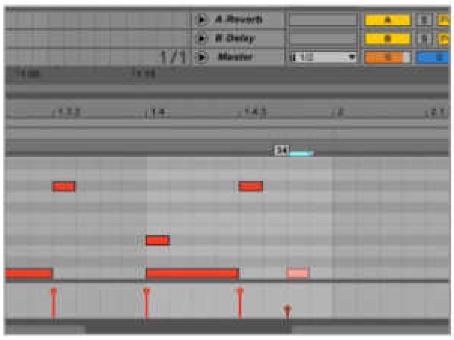
11 > Now when you play the beat back, it will loop. We no longer need the metronome, so click the **Metronome** button again to turn it off. Click the **Draw Mode Switch (Pencil)** button at the top right-hand corner of the interface. Click the **Hihat Open 909** row on **1.1.3**, **1.2.3**, **1.3.3** and **1.4.3**. (Audio: **Open hats.wav**)



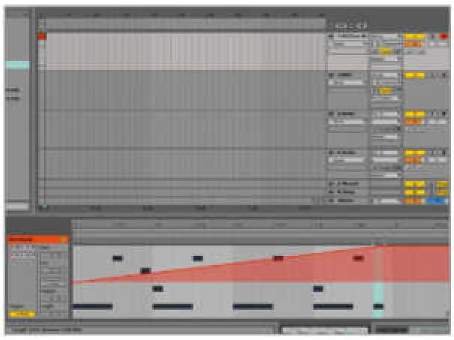
12 > We've now got the basics of a house beat, but it's pretty simple and rigid. Let's add a little swing. Press **Ctrl/Cmd+4** to turn Live's **Snap to Grid** mode off. This allows us to add notes between the lines of the grid. Add a single note on the **Hihat Closed 909** row just before the last 32nd-note of the first beat, as shown above. This subtle change adds groove to our previously plodding beat. (Audio: **Closed hat.wav**)



13 > Changes in volume are another way to add a funky human touch to programmed beats. Re-activate **Snap to Grid** with **Ctrl/Cmd+4**, and add another **Kick 909** on the last 16th-note of the bar. (Audio: **Extra kick.wav**)

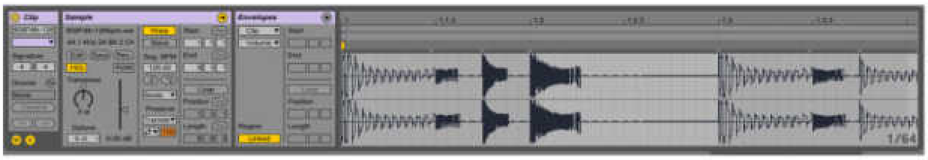


14 > Deactivate the **Draw Mode Switch** button by clicking it or pressing **Ctrl/Cmd+B**. This allows us to edit the **Velocity** of each note, which determines how hard the it 'hits'. Drag the new **Kick 909** note's **Velocity** in the MIDI Velocity Editor, and a value will appear above it. Set it to about **34**. (Audio: **Ghost kick.wav**)



15 > Other MIDI parameters can be used to affect how the part plays back. Click the **Show/Hide Envelopes Box (E)** button at the bottom left-hand corner of the Clip View. Double-click the envelope on the right to add a breakpoint. Create a couple of breakpoints and drag them around to create a ramp. On playback, you'll hear that the pitch of the beat follows the envelope we've just created. (Audio: **Pitched beat.wav**)

Warping audio clips



If you're a veteran of another DAW, such as Logic or Cubase, the way Live's audio clips work can seem a little confusing, and even counterintuitive. However, once you understand the principle behind it, you'll discover that Live really does offer a level of speed and flexibility that other software struggles to match.

Live's take on audio tracks combines sampler-like functionality with the ability to fine-tune the timing and pitch of clips. Audio clips can be set to warped or unwarped mode. In unwarped mode, the speed of clip playback is independent of Live's project tempo. You can transpose clips up and down (making it possible to create melodies without using a sampler instrument), which will affect their speed, much like changing the pitch on a record player: pitch the clip up and it plays back more quickly, down and it plays back more slowly.

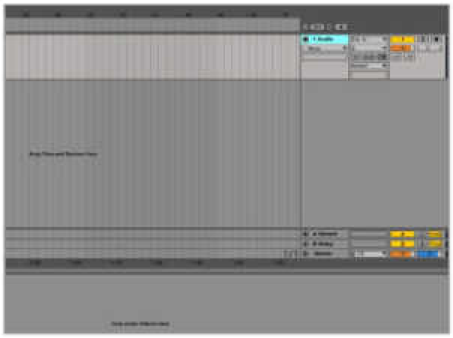
In warped mode, the speed of playback is independent of any transposition, and you specify it relative to the project tempo. This makes it possible to synchronise loops recorded at different tempos and have them stay in sync when you change the project tempo in real time! It's even possible to adjust the timing of the audio on as small a scale as you like, making it easy to tighten up loose parts or create crazy, glitched-out FX. These changes can be made in real time using the Clip View, which makes throwing sounds together a fast, fun process.

The ability to warp audio files is particularly important for DJing. By pre-warping your tracks, you can ensure that they remain perfectly in time throughout your set – see *DJing with Live* on p45.

In this walkthrough, we'll show you how to warp a beat from scratch using both automatic and manual warping, and demonstrate the looping and transposing of warped clips.

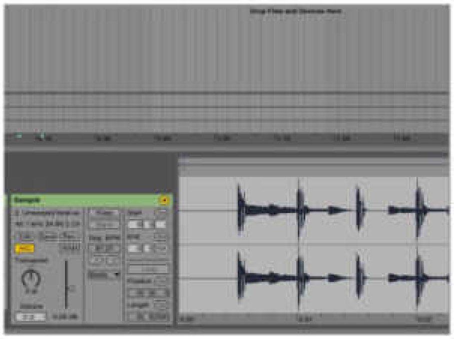
> Step by step

3. Syncing a loop to host tempo with Live's warping functionality

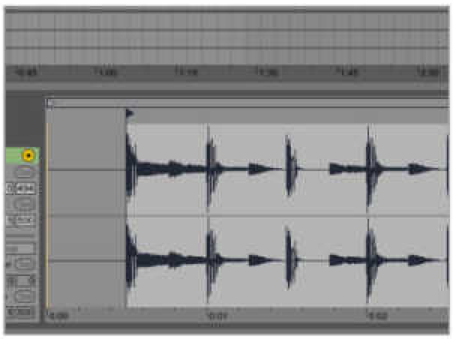


1 > Open Live and press **Tab** to switch to the Arrangement view. Let's tidy the default arrangement up by getting rid of the tracks we don't need. Click the first track's Track Title Bar to select it, then press **Delete** three times to do away with everything apart from the single remaining audio track.

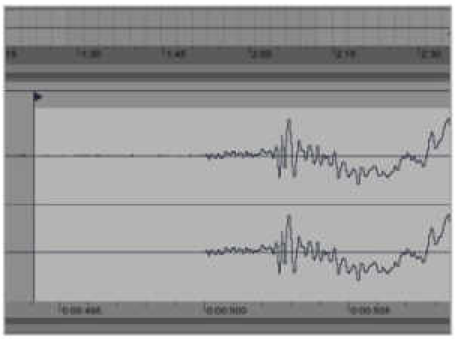
2 > In the **Tutorial Files** folder is an audio file called **Unwarped Beat.wav**. Drag it from Live's Browser or the Finder/Explorer onto the audio track. Double-click the clip on the audio track to bring it up in the Clip View. You'll notice that the **Warp** button is on, because Live has already attempted to warp the clip.



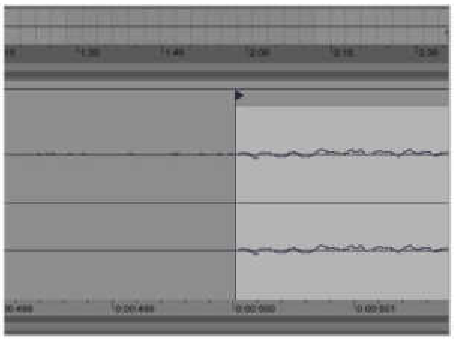
3 > Live does a great job of warping most audio automatically, but knowing how to do it manually is one of the most important aspects of using the software. Click the **Warp** button to unwarp the clip, and press the **Spacebar** to hear it in its unwarped state.



4 > The clip starts with a brief silence, followed by a two-bar beat. Drag the triangular Start Marker at the top left-hand corner of the Sample Display to the right so that it sits at the start of the first beat. Now, when you press the **Spacebar**, you'll hear the beat play back immediately.



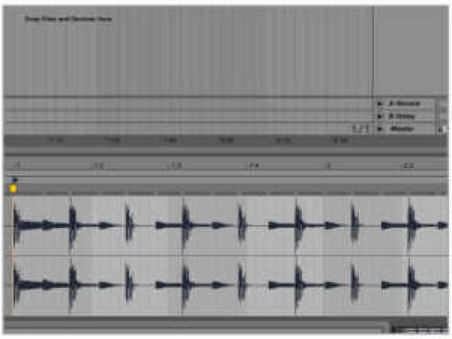
5 > When warping, you need to be as accurate as possible with your marker positioning or the audio won't play back with the correct timing. Zoom in on the waveform to make sure you've got the Start Marker in the right place. Drag downwards on the Sample Display on the first beat to magnify it. You'll likely notice a gap between the Start Marker and the actual start of the beat.



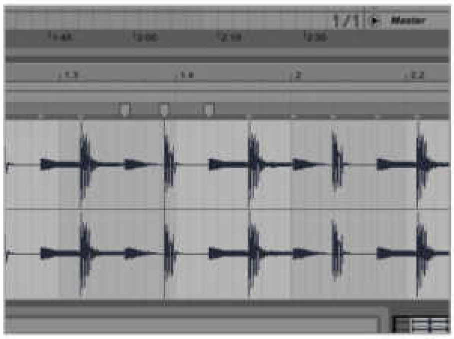
6 > Drag the Start Marker to the position shown here. When setting the start and end points of an audio clip, it's good practice to aim for points where the waveform crosses the 'zero line' – ie, the line at the centre of the waveform. This prevents the audible clicks that can occur when audio playback moves quickly from a non-zero to a zero value.

> Step by step

3. Syncing a loop to host tempo with Live's warping functionality (continued)

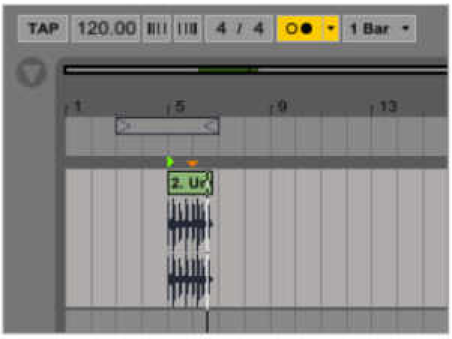


7 > With the Start Marker set correctly, we can warp the beat. Right-click the Start Marker and select **Warp From Here (Straight)**. Drag up on the Beat Time Ruler at the top of the Sample Display to zoom out and take a look at how Live's automatic warping has worked out.

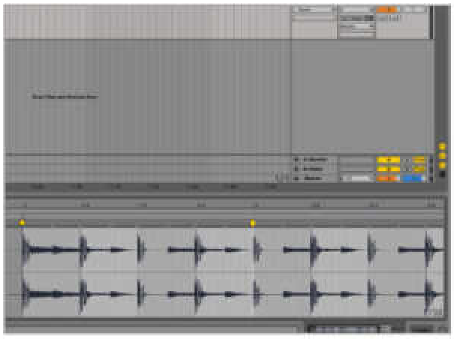


8 > You'll see that the beat divisions at the top of the Sample Display don't line up with the peaks on the waveform, which means that Live hasn't quite managed to warp the beat correctly. No matter – we can easily do it ourselves. Hover your mouse pointer over the fifth big peak (the kick at the start of the second bar) and a grey Pseudo Warp Marker will appear.

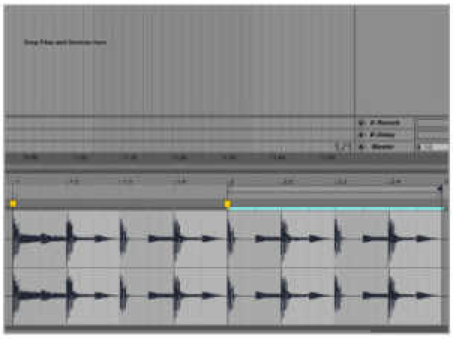
9 > Double-click the Pseudo Warp Marker and it'll turn into a yellow-green Warp Marker. Drag this to the right so that it sits at the start of bar 2. Now the first bar of the audio is warped correctly, but on playback you'll notice something odd: Live's project tempo changed when we imported the audio.



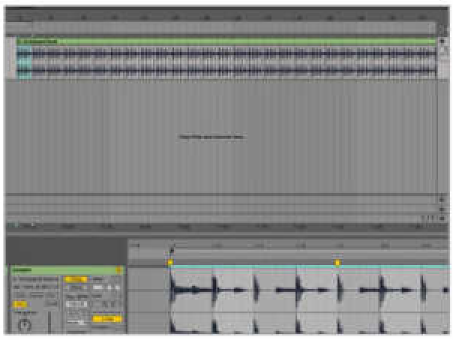
10 > Click the **Tempo** field at the top left-hand corner of the display and change it to **120bpm**. Click the **Metronome** button to activate it, and play the beat back again. You'll hear that the first bar of the beat is in time, but the second isn't. (Audio: **Partially warped.wav**)



11 > Right-click the Warp Marker at the start of the second bar and select **Warp From Here (Straight)**. Because we've warped the first bar correctly, Live can guess the tempo of the second bar more accurately and warp it properly. On playback, both bars of the beat will now be in time with the metronome. Nice! (Audio: **Fully warped.wav**)



12 > Set the end point of the clip by dragging the End Marker to the start of the third bar. Now let's loop the beat. Click the **Loop** button in the Clip View, then move the Loop Start Marker to the beginning of the second bar, and the Loop End Marker to the end of the second bar.



13 > Drag the end of the clip on the audio track out to the right and it'll extend indefinitely. Because we've only looped the second bar, the crash that plays at the start of the first bar will only play once. Alternatively, we can loop both bars simply by moving the Loop Start Marker to the start of the first bar.

14 > You can change the pitch of the audio playback with the **Transpose** knob. It's even possible to automate this to create special FX. Click the **Show/Hide Envelopes Box** button (**E**) at the bottom left of the Clip View, change the **Control Chooser** from **Volume** to **Transposition Modulation**, and click on the waveform to create breakpoints. Here, we've pitched the beat down at the end of the second bar. (Audio: **Pitched down.wav**)

POWER TIP

>Warp factor

The Clip View's **Warp Mode** parameter has a profound effect on the sound of the audio being warped. By default it's set to **Beat**, which is ideal for our example here as it preserves the punch of percussive material. However, **Beat** doesn't work well with most sustained musical material, making it sound choppy. Complex Pro is good for most things, especially vocals, as it compensates for formant shifting when transposing; but it can have an adverse affect on transients. Experiment with all the available Warp modes to find out which ones work best for any given situation.

Pack man

> Live also has its own proprietary bundle format for installing extra sound, device and patch bundles to its library: the Live Pack. Installing them is painless – simply download a Live Pack (find a large selection of free and commercial ones at **www.ableton.com/en/packs**), which comes in .alp format, and double-click it to begin installation. If Live isn't open already, it will launch and the pack will automatically install. Once installation is complete, you can find the new material located in the Browser, in **Places/Packs**.

Managing sounds and projects

Organising your data is perhaps the least glamorous aspect of music production, but it's certainly an important one: there's not much point making an amazing piece of music if you can't find it or accidentally delete it! By taking a little time to understand how Live handles projects, we can ensure that this doesn't happen and make sure that when we're collaborating with other musicians, we send them everything they need to load up a project successfully.

There are two components to a Live Project: the Project folder, and the individual Live Sets within it, which are basically Live's 'documents'. The Project folder contains all the versions of the Live Set that you've saved (saving out incrementally numbered versions is always recommended, in case you ever need to revert to an earlier one), plus related audio files, devices, and clips.

Live creates a new Project folder and stores a new Live Set there every time you save one under a new name or in a new location, *unless* you're already saving it into an existing Live Project. So, it's good practice to create a dedicated folder for all your Live Projects, then save each Project into a separate subfolder. Whenever you start a new track, save it to the root of said dedicated folder and a new Live Project folder will be created. That way, you won't end up with a single huge Live Project, which is what happens if you save all your Live Sets in the same folder!

Saving grace

Another important thing to remember (particularly when sharing projects) is to save any imported audio files into the Project folder. Live doesn't do this by default, instead referencing them directly from their original location, which can cause problems if you use sounds from a removable source such as an external hard drive or DVD-ROM.

To save all external audio files into the Project folder, select **File»Collect All and Save**. The resulting dialog will ask you to specify the locations to copy files from: Factory Packs (off by default), the User Library, other Live Projects, and other locations (all on by default). You'll rarely need to make any changes to these settings, so click the OK button to save all your imported audio to the **Samples/Imported** folder in the Live Project folder. Now you can back up your Project folder or send it to a collaborator knowing that all the audio involved will be included, and that the project can be loaded up on any system – as long as it's got the same third-party plugins installed. If you're working with someone who doesn't have a certain plugin installed, you can work around the problem by freezing the track in question before you save the Live Set and share the project. Then, simply send the Project folder and all its contents to your associate and you're good to go!

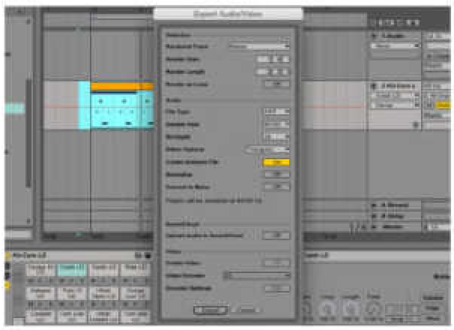
Project management is made much easier than it would otherwise be by the Collect All and Save function, which copies external files into the project folder

Lost and found

If Live can't find any of the files associated with a project, it'll display an error message and ask you to locate the files using the File Manager. Here, you can tell Live to search for the files automatically, but if you happen to know where they are, you'll likely find it a fair bit quicker to replace them by dragging them into the File Manager yourself.

Another handy logistical feature in Live is the ability to merge projects. For example, say you've created a melody that you think would work really well with a beat from another project. Simply drag the beat project's Live Set into Live with the melody project loaded and the two will be merged, combining all their devices and automation!

Audio can be exported from Live via **File»Export Audio/Video**. By default, this exports the output of the Master track at 16-bit, though there are 24- and 32-bit export options, too. If you're working on a track with someone who's using a different DAW, you can export stems by selecting the tracks you want to export and setting the **Rendered Track** to **Selected Tracks Only**.

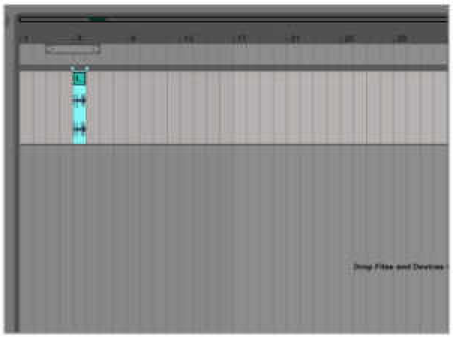


Live's Export Audio/Video dialog presents a range of useful options, including export of individual tracks and bouncing of selected tracks only

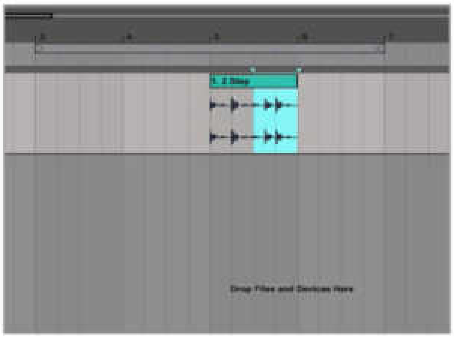
Editing in the Arrangement view

> Step by step

4. Re-arranging, consolidating and fading clips

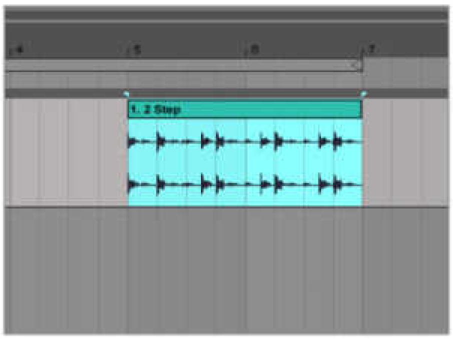


1 > Press **Tab** to switch to Live's Arrangement view, which makes editing audio and MIDI clips quick and easy. Both are edited in the same way; here we're going to work on an audio clip. Delete the two MIDI tracks and one of the audio tracks, then drag **2 Step.wav** from the **Tutorials Files** folder onto the remaining audio track.

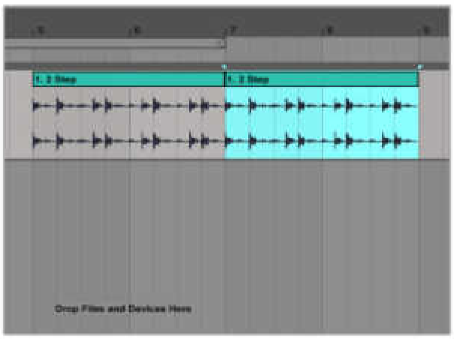


2 > The clip is looped by default, so we could drag its right-hand edge out to extend it, but let's use Live's clip editing capabilities to do something a little more interesting. Use the Beat Time Ruler to zoom in on the clip, and drag over the second half of it to select it.

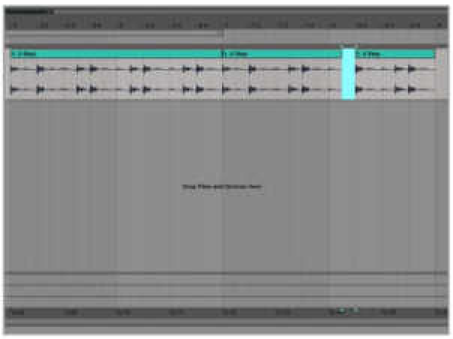
3 > With the last two beats of the bar selected, press **Ctrl**/**Cmd**+**D** to duplicate the highlighted section. Repeat this action to duplicate the two beats again, for a total of two bars. This gives us a different rhythm to what we would have got if we had simply extended the loop. (Audio: **New rhythm.wav**)



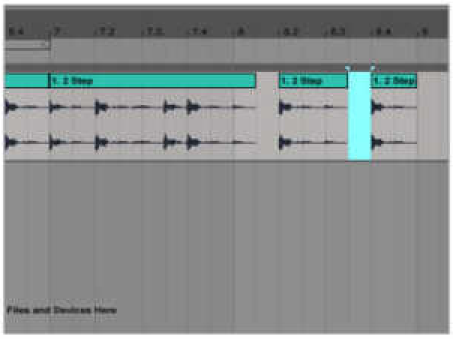
4 > The new beat comprises three separate clips, which is a little unwieldy. We can turn them into one big clip by consolidating them: select the whole two-bar beat and press **Ctrl**/**Cmd**+**J**.



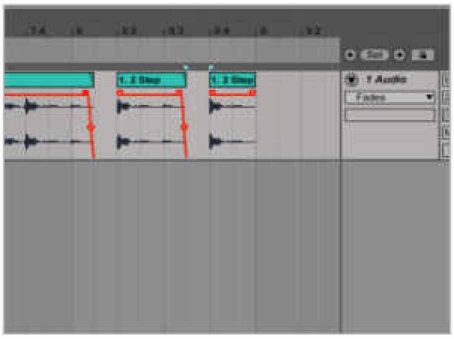
5 > As well as duplicating selected areas, you can cut, copy, paste and delete them, too. Click the clip to select it, press **Ctrl**/**Cmd**+**C** to copy it, then click the start of the bar after the beat and press **Ctrl**/**Cmd**+**V** to paste a copy onto the next bar.



6 > You can delete parts of a clip to make an 'edit'. Zoom in with the Beat Time Ruler so that the grid behind the clip displays eighth-notes, as shown above. Highlight the kick drum on the second eighth-note of the second bar, and press **Delete** or **Backspace** to delete it.



7 > Do the same with the kick drum after that. Playing the beat back, you'll notice that this creates a click before each deleted section. This is because the digital audio signal is quickly moving from a non-zero to a zero value. We can prevent this from happening using a tool particular to audio tracks: the fade. (Audio: **Clicks.wav**)



8 > In the Track Title Bar, locate the Fades/Device Chooser. Click it and select **Fades** from the menu that appears. Fade handles will appear on every audio clip. Drag the fade handles before the edits to the left slightly, to create a short fade. The edits will now play back smoothly. (Audio: **Smooth edits.wav**)

POWER TIP

>Cross talk

As well as the simple fades that we've used here, you can 'crossfade' from one audio clip to another to fade between them smoothly. This is useful when working with sustained sounds like pads. To crossfade between two audio clips, put them next to each other, make sure **Fades** is selected in the Fades/Device Chooser, and drag the fade handle of the first clip over the second. A crossfade curve will appear. If this doesn't work, the first clip may not be long enough. One way to remedy this is to activate the clip's **Loop** button in the Clip View.

Tightening tracks with quantisation

Quantisation is a complicated-sounding word, but it's nothing to be afraid of and can be your best friend when it comes to getting your music sounding tight. Put simply, quantisation is about rounding values to a nearest selected 'whole' unit of precision. Examples include rounding £9.99 up to £10, or moving a MIDI note that's slightly late onto the precise beat that it was supposed to be played on. It's possible to quantise MIDI data manually by moving it yourself, but it's often quicker to do it automatically. Ableton Live has several quantisation tools, each of which does a slightly different thing. In this walkthrough we're going to be looking at standard quantisation and record quantisation.

Live's standard quantisation functionality moves selected MIDI notes or all of the notes in a selected clip to the nearest regular time division, as specified by the user. So, let's say you want a tight 16th-note hi-hat pattern: you simply play the part in, set the Quantize resolution to 1/16 and apply the quantisation.

You can save even more time by using record quantisation: set Record Quantization in the Edit menu to 1/16 before you start recording and the part will be quantised automatically by Live the moment you press the Stop button.

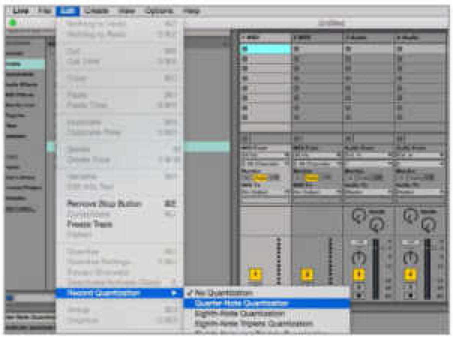
Other quantisation functions offered by Live include the ability to adjust the end points of MIDI notes as well as their start points, and control over the 'depth' or strength of quantisation applied.

Another useful tool for inputting tightly sequenced parts is MIDI step recording, which is covered briefly in *Live Secrets* on p135. You can read more on swing quantisation in *The Groove Pool* on p27, and on global quantisation in *Using*

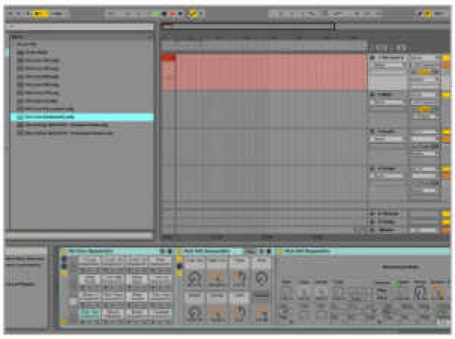
Give your drums that professional, tight feel with Live's quantisation capabilities

> Step by step

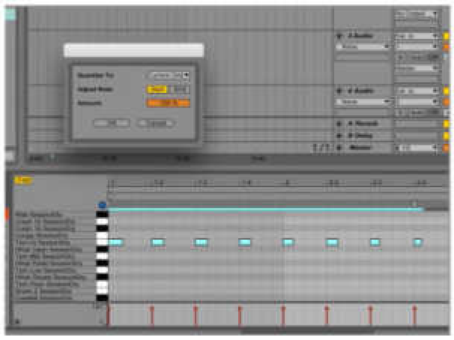
5. Quantise basics



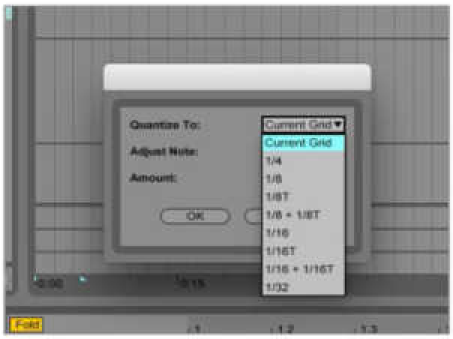
1 > By default, Live's record quantise is turned off, so when you record a MIDI part, it'll place the notes exactly as they're played. This is fine when you want a 'human' feel, but if you're after parts that are as tight as they can possibly be, record quantisation can be useful. Select **Record Quantization»Quarter-Note Quantization** from the **Edit** menu.



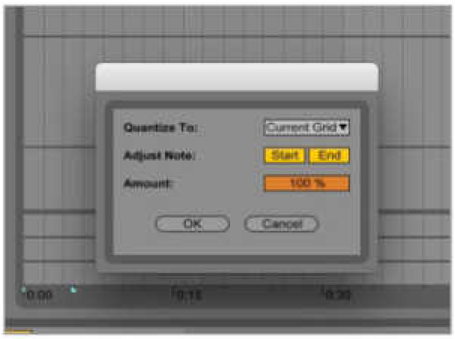
2 > Select the Arrangement view and drag **Kit-Core SessionDry** from the **Drums** category in the Browser onto a MIDI track. Activate the **Metronome**, then click **Record** and play any of the drum sounds along with it for a couple of bars. When you click the **Stop** button or press the **Spacebar**, the new notes will be quantised to the nearest beat.



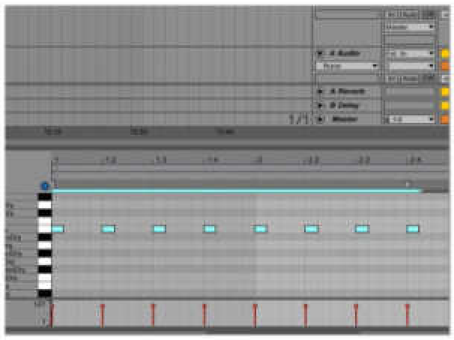
3 > Select resolutions from quarter-notes to 32nd-notes from the **Record Quantization** menu. Of course, you can also apply quantise after recording in the MIDI Editor. Double-click the clip to bring up the MIDI Editor, then click anywhere in the MIDI Editor and press **Shift**+**Ctrl**/**Cmd**+**U** to access the **Quantize Settings** menu.



4 > Here, you can select from the same range of resolutions as the **Record Quantization** menu, or choose **Current Grid**, which quantises to the selected resolution of the MIDI Editor. Live's grid resolution changes dynamically when you zoom in, so this is a convenient way to quantise to specific resolutions without having to return to the **Quantization Settings** menu.



5 > Activate the **Adjust Note: End** button, which tells Live to quantise where notes end as well as where they begin. The **Amount** parameter controls the strength of the quantisation, so values less than **100%** will retain a little looseness – ideal if you don't want your MIDI parts to sound *too* perfect.



6 > Click **OK** to close the menu and apply quantisation to the currently selected clip. To apply quantisation without opening the **Quantization Settings** menu, press **Ctrl**/**Cmd**+**U**. If nothing is selected in the MIDI Editor, this will automatically affect all the MIDI notes in the clip; if one or more notes are selected, only those will be quantised.

Turning audio into MIDI

One of the most exciting features introduced in version 9 of Ableton Live was the ability to analyse audio clips and turn them into MIDI. This powerful tool comes in three flavours: Melody, Harmony and Drums. Melody is designed to work with monophonic material – ie, single instruments, such as a trumpet or bassline. Harmony works for polyphonic material like guitar, piano or synth chords. Drums turns beats into MIDI and will even attempt to identify kick, snare and hi-hat sounds, playing appropriate samples back via a default Drum Rack.

Using Audio To MIDI is as simple as right-clicking the audio clip you want to convert and selecting the required conversion mode, although the clip must be warped correctly first – see *Warping audio clips* on p13.

Live offers a choice of three source-specific methods for converting audio into MIDI clips

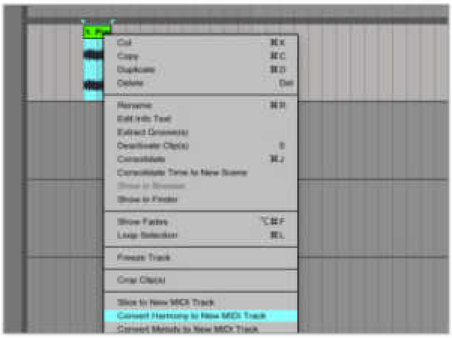
While technically very impressive, Audio To MIDI doesn't always work perfectly. There are a few things you can do to ensure optimal results, though, such as using high-quality audio sources; using parts where only a single instrument plays at a time; and using instruments with strong attack transients if possible. If errors do occur in the MIDI analysis, the easiest solution is usually to edit the part yourself, and we'll take a look at how to do that in the following walkthrough.

Live's Audio To MIDI tools have endless applications. For example, if you can't play an instrument, you can still create melody data by singing or humming into your computer's mic. You could even beatbox your own rhythms! It's great for 'borrowing' musical parts from other people's tracks, too – extremely handy for creating your own remixes.

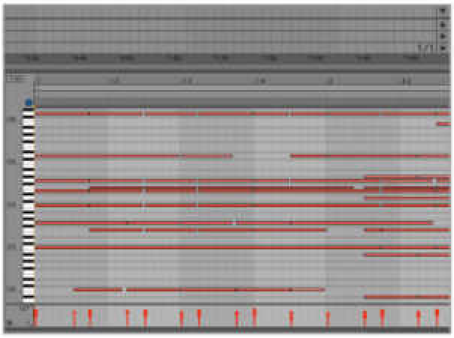
You can also use Audio To MIDI to layer audio parts on top of virtual instruments, which is what we'll do right now...

> Step by step

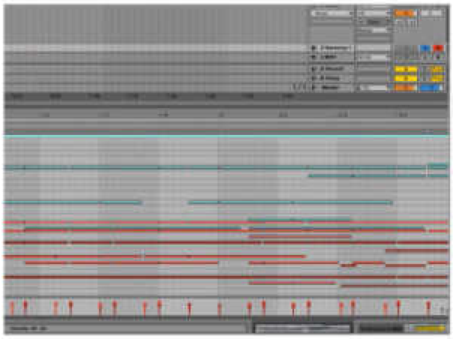
6. Doubling an audio part with Audio To MIDI



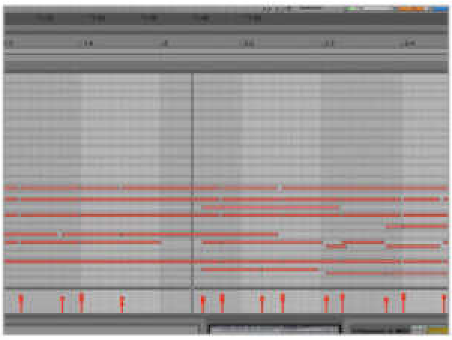
1 > Drag **Piano.wav** from the **Tutorial Files** folder onto an audio track. Audio needs to be warped before it can be analysed, but the loop we're using has already been trimmed, so Live will accurately warp it for us automatically. Right-click the clip and select **Convert Harmony to New MIDI Track**.



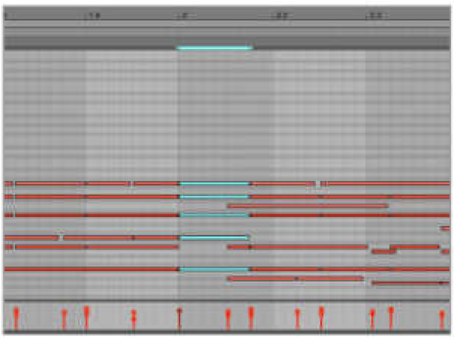
2 > Live analyses the clip and creates a new MIDI track, complete with a default piano sound with which to play it back. All very clever, for sure, but has it captured the notes correctly? Solo the MIDI track and play it back to find out. (Audio: **Initial analysis.wav**)



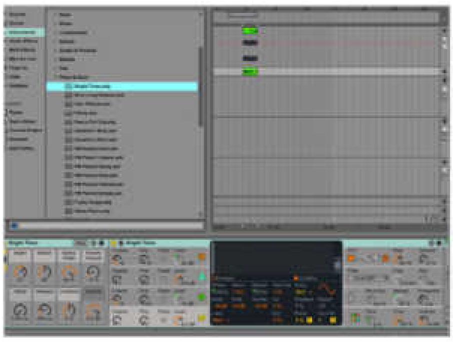
3 > While the first half of the clip has been converted correctly, there are a few issues. The main problem is that there are multiple erroneous notes at F4, G♯4, D5, B6 and D6, and a single rogue note on D♯6. The easiest workaround for this issue is to simply delete the incorrect MIDI data. (Audio: **Deleting notes.wav**)



4 > With the unwanted notes deleted, unsolo the MIDI track and play both tracks back at once to compare them. Turn the original audio track down to **-3dB** if you're having trouble making out the MIDI part. It's clear that there's another problem with Live's analysis: the chord at the start of the second bar is missing!



5 > The easiest way to deal with this is to shorten any notes from the first bar that overlap into the second, then draw in the replacement notes using the same chord at the start of the first bar. For this chord to sound consistent with the rest of the MIDI, you need to ensure its velocity is at the same sort of level in the MIDI Velocity Editor, like we have here. (Audio: **Replaced chord.wav**)



6 > Now we've got a part that sits well with the original audio, so let's try a new instrument! Select the **Instruments** tab in the **Categories** section of the Browser, then click the arrow next to **Operator** in the Browser Content Pane to show the available Operator presets. Open the **Piano & Keys** folder and drag **Bright Tines** onto the MIDI track. Finally, play the parts back to hear how they sound together. (Audio: **Layered piano.wav**)

Audio slicing and creating custom Library content

As well as its amazing ability to turn beats into MIDI clips (see previous page), Live can also automatically cut up loops so that their individual 'slices' (ie, each beat, note or chord) can be triggered via MIDI. It does this using the transient markers detected during the warping process (see *Warping audio clips* on p13), creating a Drum Rack with a Simpler or Sampler instrument for each Warp Marker or Pseudo Warp Marker. It also generates a MIDI clip that triggers these slices with their original timing, giving you a great deal of flexibility when it comes to editing and re-arranging loops.

What's more, as the audio is now being played back via Live's Simpler or Sampler instrument, it can be edited to affect the sound. The Drum Rack's macro knobs will automatically be assigned to the samplers' parameters, with the assignment depending on which of Live's slicing presets is used. These give you a variety of ways to play around with your loops, and if you find that you're not happy with the supplied presets, you can always create your own.

This leads us into the other half of this tutorial: creating your own custom Library content. Part of what makes Live's workflow so awesome is how quick and easy it is to save your own Instrument and Effect Racks into the User Library. If you create something that you think you'd like to use again, simply drag it into a folder in the User Library and it'll be stored there, ready to recall any time you like.

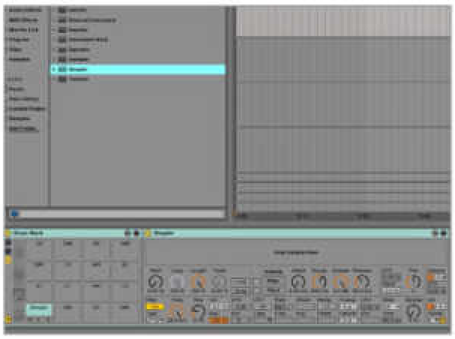
In the walkthrough below, we're going to show you how to make a custom slicing preset that's perfect for tweaking drum loops, with quick hands-on control of the slices' volume, velocity sensitivity, amplitude envelope and pitch. Let's get to it!

> "If you find that you're not happy with the supplied presets, you can always create your own"

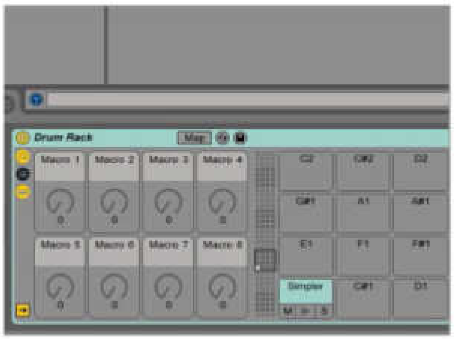
> Step by step

7. Creating a custom slicing preset and storing it in the User Library

1 > In **Categories**, click Drums, then drag **Drum Rack** from the Browser Content Pane onto a MIDI track. We're going to tell Live how we want our slicing preset to work by setting up a template Drum Rack and adding it to the **Slicing** preset folder. Because not all Live users have access to Sampler, we'll use Simpler for this.



2 > Click the **Instruments** button in **Categories**, then drag Simpler onto the Drum Rack's C1 slot. Our edited instrument will be replicated for each slice of audio every time we use the slicing preset, so it's worth taking time to get it exactly how we want it.



3 > At the top left-hand corner of the Drum Rack is the **Show/Hide Macro Controls** button. Click this and eight 'macro' knobs will appear. These are important, because we can use them to control the parameters of every instance of Simpler created by the slicing process – a lot more convenient than adjusting the parameters of each Simpler manually.

4 > On the right-hand edge of the Simpler instrument is the Volume parameter. Right-click this fader, and select **Map to Macro 1**. This sets up the macro, but it also lowers the parameter to minimum: **-36dB**. Left unchecked, this would be the default value for all Simplers created with the preset. Let's address this now.

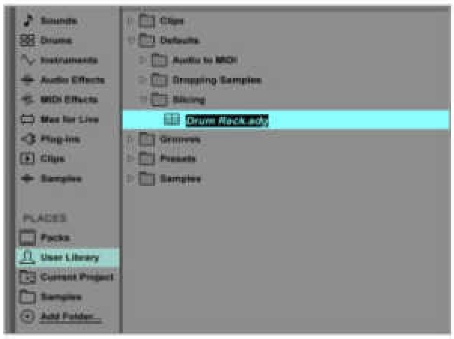
5 > Click the **Macro 1** knob and enter **-12** for its value, then press **Enter**. Setting up macros is as simple as that: pick a parameter and enter a default value. Let's set up the rest of the available macros. We want to be able to control how heavily velocity affects the volume of our slices, so right-click the **Vel** parameter below the **Volume** fader and select **Map to Macro 2**.

6 > Click **Macro 2** and set it to **35**. With the volume-related controls sorted, let's move on to pitch. It's not possible to tune a sample with just a single parameter in Simpler, because it has separate knobs for **Transpose** (semitone tuning) and **Detune** (cent tuning). Assign **Transpose** to **Macro 3** and **Detune** to **Macro 4**, and set both of their default values to **0**.

> Step by step

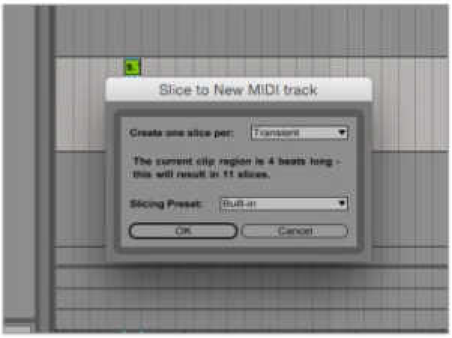
7. Creating a custom slicing preset and storing it in the User Library (continued)

7 > There are four slots left – just enough to control Simpler's Volume envelope. Assign **Attack** to **Macro 5**, **Decay** to **Macro 6**, **Sustain** to **Macro 7** and **Release** to **Macro 8**. Set the **Decay** to **600ms**, **Sustain** to **0.0dB** and **Release** to **50.0ms**. The **Attack** will be **0.00ms** by default.

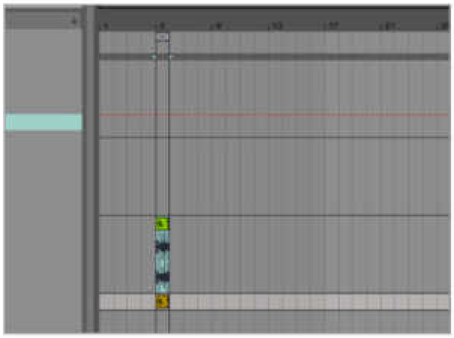


8 > Now we're ready to save the preset. In **Places**, select the **User Library**, then, in the Browser Content Pane, click the arrow next to the **Defaults** folder to see its contents. Drag your new Drum Rack into the **Slicing** folder. A preset will appear in the folder, with its name automatically selected ready for you to enter.

9 > Type 'Breakbeat' and press **Enter** to name the preset. Adding presets works like this for everything in the User Library: just drag whatever you've created into the appropriate folder and name it. Now we're ready to test the preset out. Drag **Vintage break.wav** from the **Tutorial Files** folder onto an audio track.



10 > Live automatically warps the break, so it's ready to slice right away. Right-click the audio clip and select **Slice to New MIDI Track**. A menu will appear, asking how you want the audio to be sliced. By default, one slice is created per transient, which is perfect – all we need to do is to tell Live to use our new slicing preset.



11 > Click the **Slicing Preset** menu and select **Breakbeat** from the list, then click **OK**. A new MIDI track will appear with a Drum Rack on it. Solo the MIDI track, then click the MIDI clip and press **Ctrl/Cmd+L** to loop around the clip.

12 > Press the **Play** button to listen back to the sliced audio. If you can't see the slices' pads, click the light grey boxes in the Drum Rack's Pad Overview. As the beat plays back, each slice's **Preview** button will illuminate to show that the slice is playing.

13 > Adjustments made to the Drum Rack's macros affect the Simpler instruments on every pad. For example, turning the **Sustain** macro down to **-inf dB** will result in a tighter drum sound, and we can further tweak the tightening effect by turning the **Decay** macro up or down. (Audio: **Tighter beat.wav**)

14 > We can also independently process specific slices, of course. For example, click **Audio Effects** in **Categories**, then drag **Reverb** onto Slice 3's pad to add a reverb effect to just the first snare hit. This technique is particularly useful with EQ, enabling you to high-pass each slice with its own cutoff frequency – great for cleaning up classic funk breaks. (Audio: **Reverb slice.wav**)

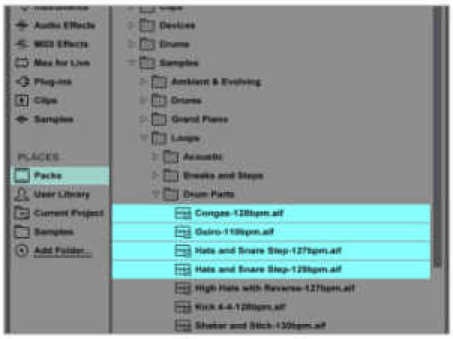
POWER TIP

>Trigger finger

A small but important option hidden away at the bottom right-hand corner of the Simpler interface is the **Retrigger** button – a tiny box with an 'R' in it. With this active (yellow-green rather than grey), whenever a new note is triggered, the previous note will be stopped dead. This doesn't make a huge difference most of the time, but with long release times, the effect will be more obvious. So, if you're using loops with sustained sounds, like synth chords, try experimenting with the **Retrigger** button to see which mode works best for you.

> Step by step

8. Recording an arrangement live with the Session view



1 > Start a new project – it will open in Live's Session view by default. In the Browser, select **Places**, then **Packs**, then click on the arrow next to **Core Library**. Open **Samples/Loops/Drum Parts** and highlight the first four samples by clicking **Congas-128bpm.aif**, holding the **Shift** key, and clicking on **Hats and Snare Step-129bpm.aif**.

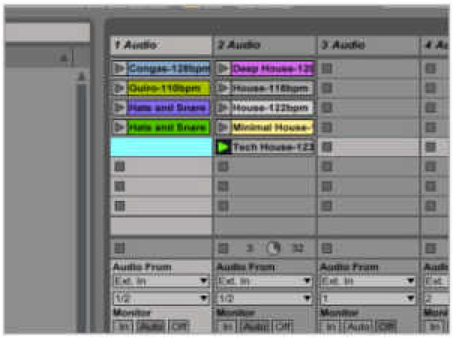
2 > Drag the samples to the top of the first track in the Session view. Don't worry about it being a MIDI track – Live will convert it to an audio track and change the project tempo to that of the first sample: **128bpm**. Turn on the metronome and click the **Clip Launch** button on the **Congas-128bpm clip**. These clips have already been warped, so they'll play back in time with the metronome.

3 > Only one clip can play on a channel at a time. Click the **Clip Launch** button on the **Hats and Snare Step-129bpm** clip and it will start playing on the downbeat of the next bar. You can change when new clips start playing with the **Quantization** menu. Click it and select **1/4**.

4 > Now, when you click a **Clip Launch** button, the clip will begin playing at the start of the next beat. To stop a clip, click any of the square **Clip Stop** buttons on empty clips, or the **Clip Stop** button on the Track Status Display. You'll notice that the metronome keeps playing: you've only stopped the clip, not Live.

5 > To stop Live playing, click the **Stop** button on the Control Bar, or press the **Spacebar**. Set the **Quantization** menu back to **1 Bar** and turn the metronome off. Now open the **Electronic** folder in the Browser Content Pane and drag the first five clips to the top of the second track.

6 > Click the **Clip Launch** button on the **Congas-128bpm** clip, then click the **Clip Launch** button on the **Deep House-120bpm** clip. At the start of the next bar, the **Deep House** beat will start playing in time with the **Conga** loop. Stop the **Conga** loop by clicking one of the **Clip Stop** buttons on the first track, and the **Deep House** beat will carry on playing.



7 > At the right-hand side of the Session view are the **Scene Launch** buttons. Click the one on row 3 to launch all the clips on that row together. This triggers any **Clip Stop** buttons in the Scene, too – click the **Scene Launch** button on row 5 and the clip playing on the first track will stop. To prevent this happening, select an empty Clip Slot and press **Ctrl/Cmd+E**, which will remove the **Clip Stop** button.

8 > Click the **Stop All Clips** button on the Master track to kill all playback, then click the **Record** button and have a jam, triggering a selection of clips on each channel. When you're done, click the **Stop** button, then press **Tab** to toggle to the Arrangement view. Your jam has been recorded, but all the clips are greyed out. To activate them, click the orange **Back to Arrangement** button to the right of the scrub area.

POWER TIP

>Scene it

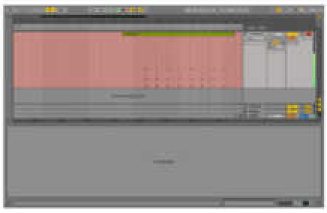
When you're using Live to generate new ideas or as part of a live performance, things in the Session view can get complicated quickly. Thankfully, the **Capture and Insert Scene** function can help you stay on top of it all. When used (either from the **Create** menu or by pressing **Shift+Ctrl/Cmd+I**), Live inserts a new Scene below the current selection with copies of the currently running clips in it, and launches it immediately. So, whenever you come up with a cool combination of clips, make sure you Capture and Insert Scene so that you can return to it.

MIDI overdubbing

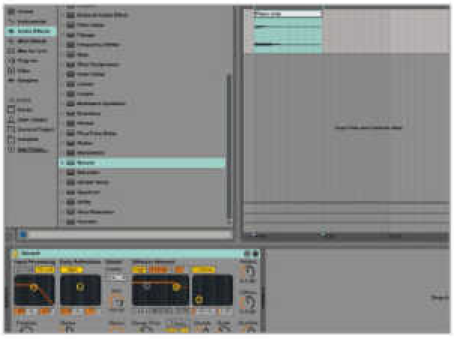
Live's MIDI tracks can be punched into and out of just like audio tracks, but they also have an additional recording mode: overdub. To activate overdubbing, click the MIDI Arrangement Overdub button (the + symbol) to the right of the Record button. Now when you record onto a MIDI track, rather than replacing any MIDI data already there, you'll add to it. This can be useful when building up complex drum parts, layering melodies over the top of chords, or creating fills. To turn off overdub mode, simply click the button again.

Looping a section is a cool way to build up a complex pattern layer by layer. If you want to try out a new musical idea while you're recording, click the Record button. Live will keep playing the looped section, but won't record anything until you click the Record button again.

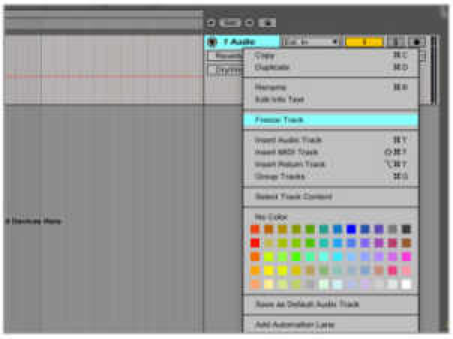
Also on the transport bar is the Re-Enable Automation button. Whenever you manually adjust an automated parameter, the automation is overridden, but clicking this button reverts the tweaked control to the automation in the Arrangement or Session clips. To find out more about automation, see *Automating parameters* on p26.



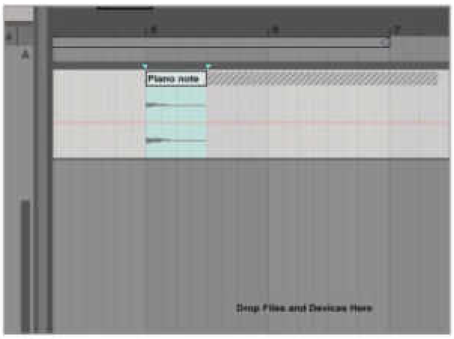
> Step by step 9. Freezing and flattening tracks



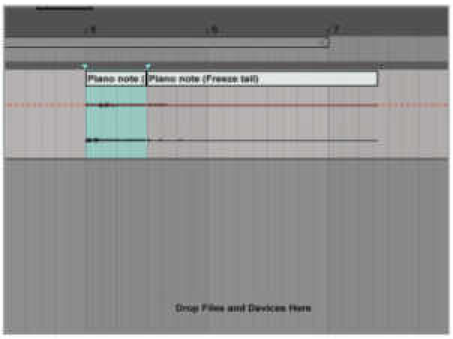
1 > Live can render MIDI instrument and effects-processed audio tracks as audio behind the scenes – useful for reducing CPU usage, sharing projects with someone who doesn't own the same plugins as you, and 'resampling' sounds of your own making. Start by dragging **Piano note.wav** onto an audio track, followed by **Reverb** from the **Audio Effects** folder.



2 > Turn the Reverb's **Decay Time** up to **4000ms** (4 seconds) and set the **Dry/Wet** mix to **50%**. This gives the piano note a long tail, lasting for about two bars. Right-click the Track Title Bar and select **Freeze Track**.

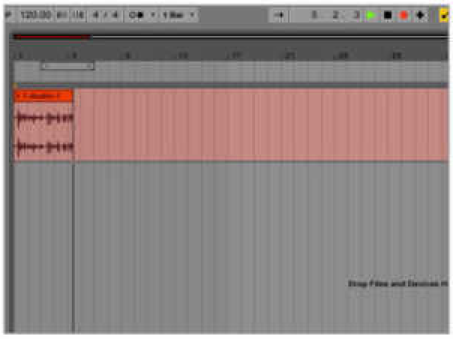


3 > Once the operation is complete, the Reverb effect is greyed out and chevrons on the audio track signify a Freeze Tail. Freeze Tails comprise any audio beyond the end of an audio clip's end point, usually resulting from reverb or delay effects.

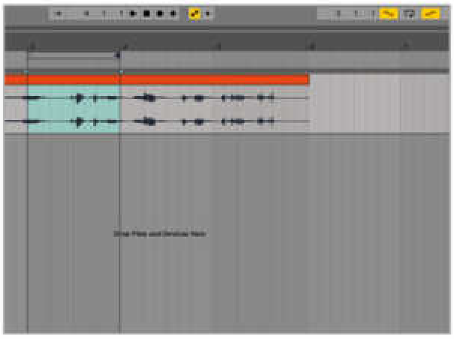


4 > Because the track is frozen, we can't change its effect settings, but we can still copy, paste and delete the audio clip and its Freeze Tail. We could revert back to the 'live' version by right-clicking the Track Title Bar and selecting **Unfreeze Track**, but by selecting **Flatten** instead, we replace the 'live' clip with the rendered audio, meaning we can process it as we would any other audio clip.

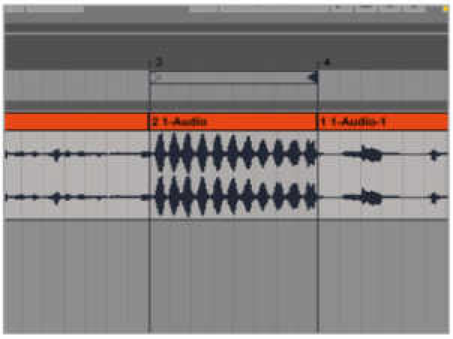
> Step by step 10. Punch-in/punch-out recording



1 > When recording, you'll sometimes want to replace only a section of a track rather than re-recording the whole thing. This is where Live's punch-in and punch-out functions can come in handy. Click the **Arm Arrangement Recording** button on an audio track, then click the **Record** button and record a few seconds of your voice, an instrument or whatever else you like.



2 > Press the **Stop** button when you're done. Now let's replace part of the recording. Click the **Punch-In** switch to activate it, and move the Loop Start/Punch-In Point to the middle of the audio clip you just recorded. Next, click the **Punch-Out** switch, and move the Loop End/Punch-Out Point to a point after the Loop Start/Punch-In Point.



3 > Press the **Stop** button to return the song position to 1.1, then press **Record** to begin recording. Only the material between the Loop Start/Punch-In Point and the Loop End/Punch-Out Point will be replaced by the new recording – everything else will be left as it is.

The Track Mixer

Live's mixer is straightforward but powerful, making light work of everything from basic mixing (ie, setting the level and pan position of each track) to complex audio routing. Although the mixer panels look a little different in the Arrangement and Session views, aside from the Session view's Crossfader Section (see *DJing with Live* on p45) and Peak Level display (which is reset by clicking it), their functionality is identical. So, it's possible to create and mix tracks entirely in the Arrangement view, though some may prefer the more traditional horizontal mixer arrangement offered by the Session view.

If you've never used a real or virtual mixer before, Live's might seem a little intimidating. Thankfully, though, it's very sensibly designed, and while it may appear inscrutable at first glance, it really couldn't be easier to use. The key to overcoming any fear of real-life mixing desks is to remember that all of those many channels are functionally identical, and that's as true for Live's mixer as it is for that of any other DAW. Both audio and MIDI channels have the same controls, and you'll get the hang of them in no time, we promise.

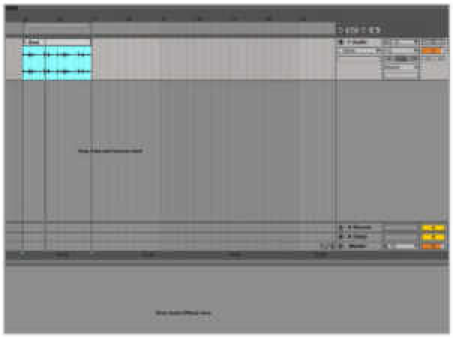
We've seen how the Arrangement view's Arm Arrangement Recording button works in *Sequencing your first beat with a MIDI track* on p11, and the only difference between that and its Arm Session Recording equivalent is that you have to click the Record button on a specific clip to record anything in the Session view.

In the step-by-step walkthrough below, we're going to show you how to use the mixer's Activator, Volume, Pan, Solo/Cue, monitoring and routing controls, but not the Sends. You can find specific instructions for these in *Sends and returns* on p28.

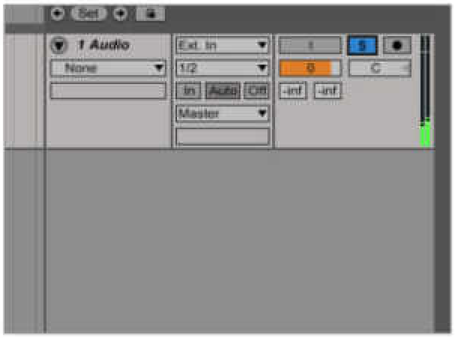
Live integrates the functionality of a hardware mixing desk directly into its Session and Arrangement views

> Step by step

11. The ins and outs of Live's mixer

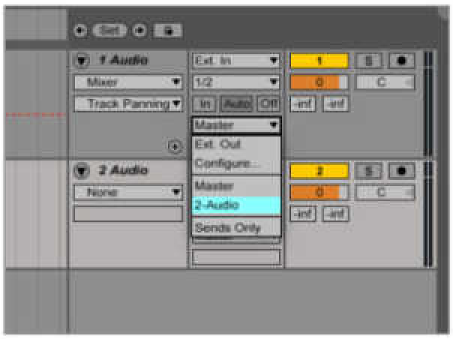


1 > In the Arrangement view, drag **Beat.wav** onto an audio track. With the clip selected, press **Ctrl/Cmd+L** to set Live's Loop Start and Loop End points around it. Let's start with the basics. Click **Play**, then click the track's **Track Activator** button. This will deactivate the track, silencing it.

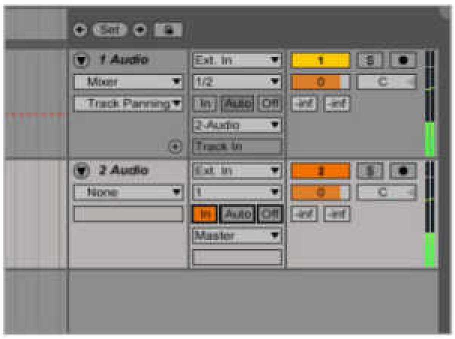


2 > Click the **Solo/Cue** button. The soloed track can now be heard again, even though it's deactivated. Solo and arm multiple tracks simultaneously by holding **Ctrl/Cmd** and clicking their **Solo/Cue** and **Arm Arrangement Recording** buttons. Deactivate the **Solo/Cue** button, and click the **Track Activator** button again to reactivate it.

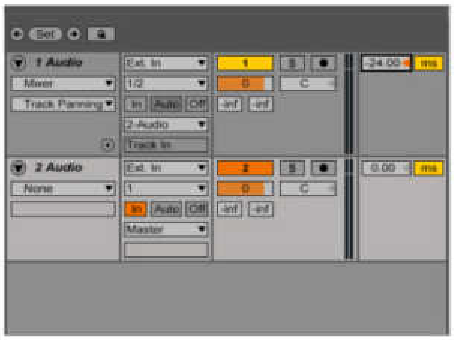
3 > Drag down on the **Track Volume** fader to reduce the track's volume level. To return it to its default setting of **OdB**, select it and press the **Delete** key. This works for all of Live's parameters, in fact, including those of third-party plugins in the Device View. Sweep the **Track Pan** to move the position of the track around in the stereo panorama. Press **Delete** to reset it when you're done.



4 > Press **Ctrl/Cmd+T** to create a new audio track/channel. We're going to route our existing audio track to this new one. On the original track, click the **Output Type**, and select **2-Audio** from the menu that appears. Routing a track is as simple as that, but you won't actually hear anything on playback yet.



5 > This is because we haven't turned monitoring on for the new track yet. Click the monitoring mode button that reads **On**, and when you play the project back, you'll hear the beat again. You can route an unlimited number of sources to a track in this fashion, enabling busses to be set up without using groups (see *Working with grouped tracks* on p42).



6 > A track can also be set to **Auto** monitoring mode, which activates monitoring only when the channel is armed – useful when recording. Finally, click the **Show/Hide Track Delays** button – the small 'D' at the bottom right of the Arrangement. This brings up the **Delay** setting for each track, which can be set to samples or milliseconds and adjusted to push/pull the track back slightly in time.

Waldemar H. aka DJ Val de Mossa, DJ department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Automating parameters

Back in the days before automation, real-time changes to mixer settings had to be performed as the master was being recorded. More complex pieces could require dozens of tweaks, sometimes necessitating pages of notes and even multiple sets of hands to get the knobs, faders and buttons into the right positions.

These days, our DAWs can automate parameter changes to a level of precision that would have been unimaginable a few decades ago. This enables us to create insane effects and sophisticated mix transitions, so getting your head around automation is a must if you want to make the most of your music.

Some software makes creating or recording automation a tricky and long-winded process, but not Ableton Live – the way it implements automation is clean, clear and sophisticated. Rather than having you search through lists of automatable parameters like some DAWs do, Live will automatically bring up the automation envelope for any selected parameter.

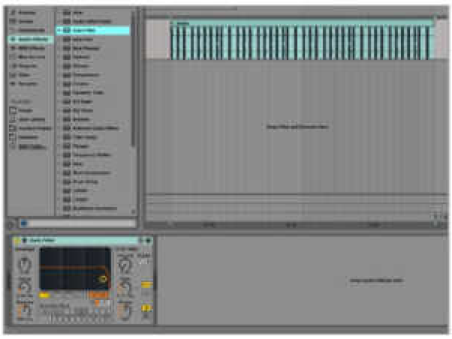
Automation can be drawn in in Draw mode – meticulous placed manually, breakpoint by breakpoint – or recorded in real time by adjusting parameters on-screen or via a MIDI controller using Live's MIDI mapping capabilities (see *MIDI and key mapping* on p43).

In this walkthrough, we'll demonstrate the automation of a single parameter, but generating more involved automation with multiple parameters is aided by Live's ability to display several automation envelopes at once. To use this feature, click the + button at the bottom of the Track Title Bar – the currently activate envelope moves to a dedicated automation lane below the track display. You can add as many of these lanes as you like, making it easy to edit and keep track of all the automation in a project.

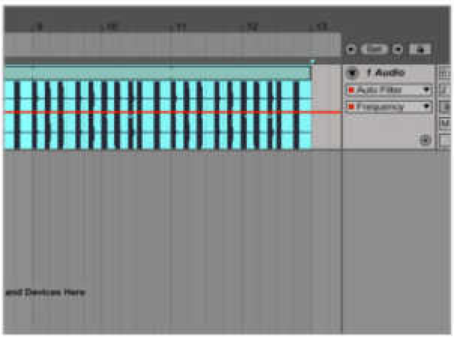
"Getting your head around automation is a must"

> Step by step

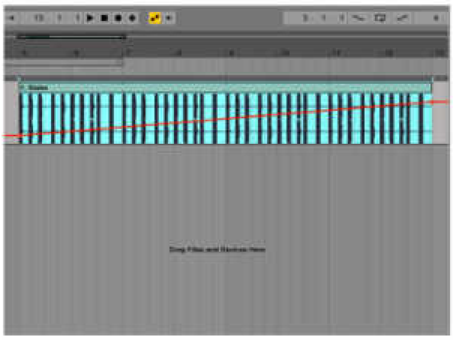
12. Creating and editing automation envelopes



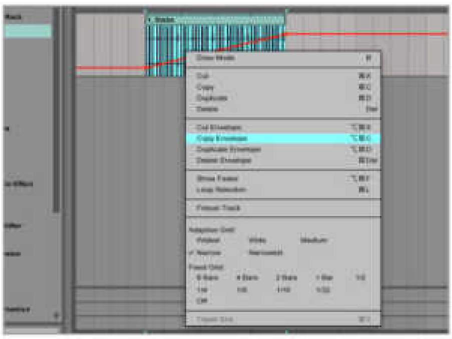
1 > In the Arrangement view, drag **Stabs.wav** onto an audio track. Live automatically loops the audio clip, so we can drag its right-hand edge to elongate it. Drag it out so that the clip lasts for eight bars. Add **Auto Filter** from the **Audio Effects** folder to the track.



2 > Click on the **Filter Cutoff** control, which will be set to **12.5kHz** by default. The audio track's **Fades/Device Chooser** and **Automation Control Chooser** will be automatically set to **Auto Filter** and **Frequency** respectively, and a red automation envelope will appear on the track. To add a breakpoint to the envelope, click it.

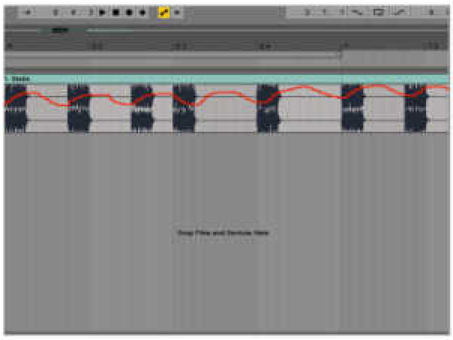


3 > To move a breakpoint, drag it around. Copy the upward ramp movement shown here to create a filter sweep. Hold **Shift** and drag one breakpoint over another to delete it; hold **Ctrl/Cmd** to fine-tune a breakpoint's horizontal position; and move multiple breakpoints at once by dragging over an area of the track to select it, then moving any of the breakpoints in the selection.



4 > Copying a clip also copies its automation, although you can lock all the automation in a project with the **Lock Envelopes** button. This prevents automation from being affected when you edit clips. You can edit automation but not the associated clip by selecting an area, right-clicking and selecting **Cut**, **Copy**, **Delete** or **Duplicate Envelope**.

5 > Automation envelopes can be curved to help fades and parameter modulation sound more natural than they often do with simple linear movements. To create an automation curve, hold the **Alt** key and drag on the envelope between two breakpoints.



6 > To record automation, click the **Record** button (you don't need to arm the track first) and adjust a parameter in real time. Manually draw in automation by clicking the **Draw Mode** switch or pressing **Ctrl/Cmd+B**. Recording or drawing in automation is usually preferable to adding breakpoints manually when you require smooth, complex movements.

The Groove Pool

Generally speaking, people enjoy listening to music and bands that sound 'tight' – ie, all keeping to the same rhythm, rather than individual members pushing ahead or falling behind. Quantisation can be useful for getting a tight sound (see *Tightening tracks with quantisation* on p17), but creating a satisfying rhythm isn't always as simple as rigidly quantising everything. In fact, what gives most popular music its foot-tapping quality is 'swing'. This involves playing certain notes – usually eighth- or 16th-notes – slightly late. The practise originates in jazz and can be heard in everything from funk and hip-hop to house and DnB. It's swing that gives Double 99's garage classic *RIP Groove* its infectious rhythm, while The Honey Drippers' much-sampled *Impeach the President* is practically swimming in it.

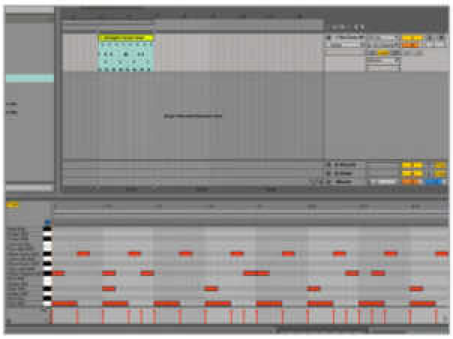
While you can funk up your tracks with swing by playing it in yourself or editing your MIDI or audio parts manually, Live features a built-in solution that makes managing a project's rhythmic structure a relatively painless process: the Groove Pool. Live 9 lets you extract the timing data from a MIDI or audio clip and turn it into a 'groove' that can be applied to any clip in the project, warping its timing to make it fit rhythmically with the groove's source clip. Live's Core Library includes dozens of prefab grooves, including some based on settings from classic hardware such as Akai's MPC range.

In this walkthrough, we'll show you how easy it is to apply grooves from Live's Library and extract your own from clips. Remember, grooves don't just work with beats, but any audio or MIDI clips, so if you're after the tightest sound possible, make them a priority.

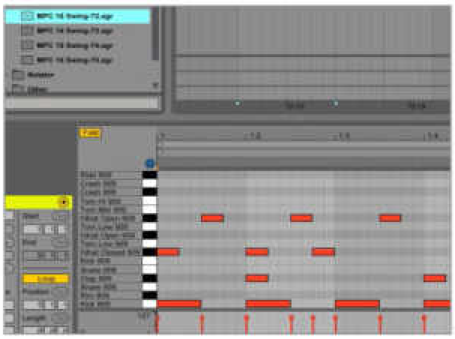
Like Akai's legendary MPC series, Live can add funky swing to sequences

> Step by step

13. Creating and applying grooves

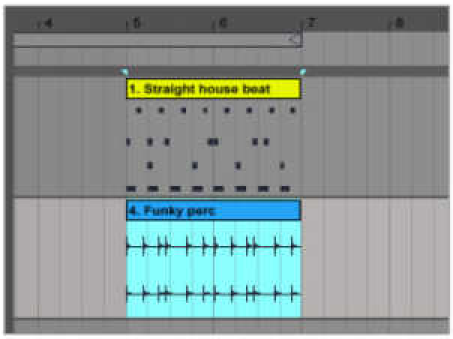


1 > Drag **Straight house beat.mid** onto a MIDI track in the Arrangement view. Live will ask you if you want to import its tempo and time signature data – you don't. Open **Drums** and drag **Kit-Core 909** onto the track, too. Double-click the MIDI clip and you'll see that the closed hi-hats are rigidly quantised to 16th-notes, making them sound very 'straight'. Let's loosen them up with a touch of swing.

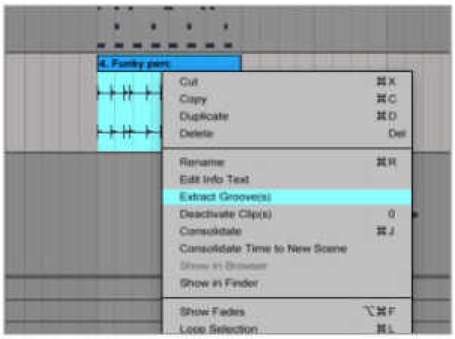


2 > In **Places**, open **Packs/Core Library/ Swing and Groove**; this is Live's library of groove templates. Let's go for a classic house feel. Open the MPC folder and drag **MPC 16 Swing-72** onto the MIDI clip. This particular swing groove affects all odd-numbered 16th-notes, and on playback you'll hear that the beat is now well and truly funked up. (Audio: **MPC house.wav**)

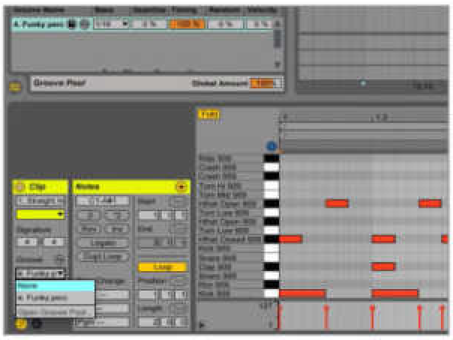
3 > If you're having trouble hearing the difference, open the Groove Pool by clicking the circular 'wavy lines' button below the Browser. Turn the **Timing** down to **0%**, then back up to **100%** as you play the beat back. Note that applying the groove doesn't actually change the position of the notes in the MIDI clip, just how they're played back.



4 > Click the **MPC 16 Swing-72** groove in the Groove Pool, and press **Delete** to remove it from the project. This returns the beat to its original rigid feel. Now drag **Funky perc.wav** onto an audio track, and listen to how it sounds with the beat.



5 > The timing of the loose percussion loop doesn't sit too well with the straight house beat. Let's fix that. Right-click the audio clip and select **Extract Groove(s)**. Live will take a moment to analyse the clip, and once it's done, a new groove will be added to the Groove Pool.



6 > Drag the new groove onto the MIDI clip. This applies the groove from the percussion clip to the MIDI notes, so that the beats sit together in a much more satisfying way. To stop the groove from affecting the MIDI clip without removing it from the pool, set the **Groove** parameter in the Clip View to **None**. (Audio: **Perc groove.wav**)

Sends and returns

The technique of using return channels dates back to the days of analogue mixers and outboard gear, when having dedicated effects units on every channel was a logistical impossibility. To share a single effect unit – typically a reverb or delay – between multiple channels, the engineers would put it on its own dedicated 'return' buss channel, which meant they could route (or 'send') as many channels as they liked to the effect together. Of course, using a single unit meant it was only possible to use one effect setting at a time per return channel to process all those tracks, but it was at least possible to control the amount of effect applied to each track by adjusting its send level.

Returns had certain other advantages, too. For one thing, they made it possible to process the effected signal further without altering its source. This meant that the producer could route a bassline channel to a reverb on a return, then place a high-pass filter after the reverb to stop its low end from clogging up the mix, without taking the low end out of the unprocessed bass.

Another useful technique made possible by the return channel is what's known as 'spin delay'. This involves turning the send level to a delay plugin up at a particular moment in order to apply delay to a specific part of a track – the end of a sung phrase, for example.

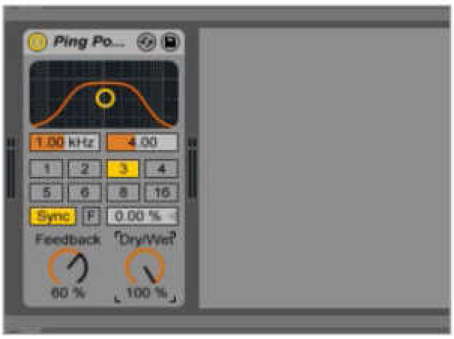
These days, as the processing power of home computers grows ever greater, we don't have to worry so much about using too many effects, and it's totally feasible to use dedicated reverbs and delays on a huge number of channels. However, thanks to its unique capabilities, the return channel is still a wholly relevant inclusion in all DAWs.

Live can host up to 12 return channels, though only the most complex and involved projects would require that many. In the walkthrough below, we'll show you how to set them up, automate them for spin delay effects, and even route them back into themselves to create feedback loops.

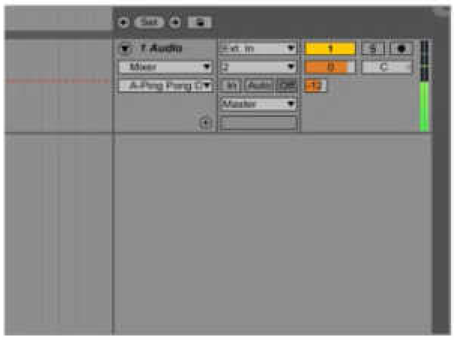
> Step by step

14. Feedback effects with send and return channels

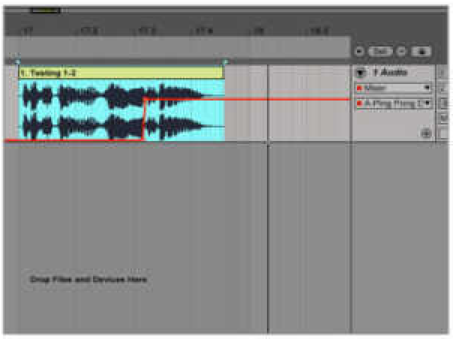
1 > Drag the **Testing 1-2.wav** vocal sample onto an audio track in the Arrangement view. Live's default project includes two return tracks with effects already loaded, but we're going to make our own from scratch. Click the return tracks' names and press **Delete** to remove them. Select **Create»Insert Return Track**, or press **Ctrl/Cmd+Alt+T** to add a new, empty return track.



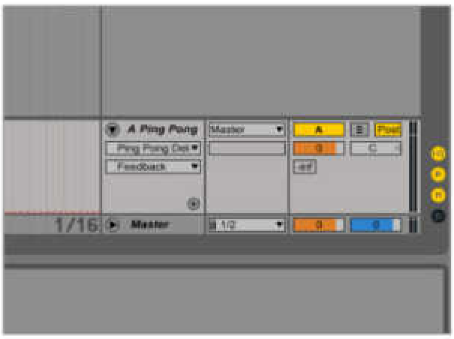
2 > Drag **Ping Pong Delay** from the **Audio Effects** folder onto the return track. Because we're going to hear the unprocessed vocal directly from the audio track, we don't need to have any 'dry' signal coming from the send, so set the Ping Pong Delay's **Dry/Wet** knob to **100%** – ie, fully 'wet'.



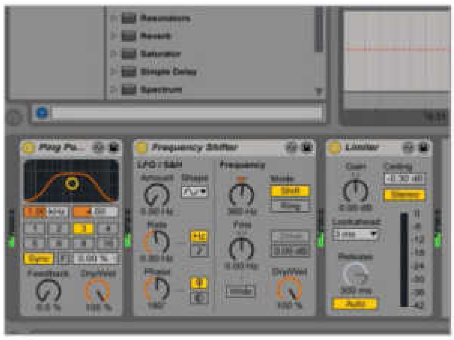
3 > In the audio track's Mixer section, you'll see a small horizontal fader, which controls the send level to the return track. Turn this up to **-12dB**, and when you play the vocal sample you'll hear the delay. Let's apply the delay to just one specific part of the vocal. (Audio: **Full delay.wav**)



4 > The send level's automation envelope will have appeared automatically when you clicked its fader. Click two different points on the envelope to create two breakpoints, and move them into the positions illustrated above. This sets the send level to **-inf dB** until the word "two", at which point it jumps to **-12dB**. As only the last word is delayed, we end up with a much cleaner-sounding effect. (Audio: **Clean delay.wav**)



5 > The Ping Pong Delay has built-in feedback, but we can also set the return to feed back into itself. So, rather than having feedback in just the delay effect, we can create our own custom feedback loop, involving as many effects as we like! Turn the Ping Pong Delay's **Feedback** down to **0%**, then click the arrow button on the return track to unfold it. The send fader is disabled as a safety precaution to prevent excessive feedback.



6 > Right-click the fader and select **Enable Send**. The send level now controls the amount of feedback; set it to **-6dB**. Drag a **Frequency Shifter** and a **Limiter** from the **Audio Effects** folder to the return track. Set the Shifter's **Frequency** to **360Hz**. Now, on playback you'll hear the return track's signal feed back into itself, giving us both feedback delay *and* feedback frequency-shifting effects. (Audio: **Frequency shifting.wav**)

Audio Effects

Let's take a break from the walkthroughs with an A-to-Z tour of Live's built-in devices, starting with its extraordinarily powerful and creative array of audio effects processors

AMP

Emulates seven classic guitar amplifiers. It's designed to work with Cabinet, which should come after it in a device chain if your goal is to achieve authentic results with guitar. Amp even simulates the fixed amount of electricity available to real-life amplifiers, with adjustments to one parameter having an affect on the others.

AUDIO EFFECT RACK

A container for constructing 'aggregate' devices from other effects that allows multiple signal chains to be established for parallel processing, and can even nest other Audio Effect Racks inside itself. If you find yourself commonly using a particular set of effects, you can speed up your workflow by putting them in an Audio Effect Rack and saving it as a Library preset – see *Audio slicing and creating custom Library content* on p19.

AUTO FILTER

Analogue filter emulation with low-pass, high-pass, band-pass and notch modes. Features an envelope follower and LFO modulation for dynamic cutoff frequency movement. The effect actually has one LFO for each channel of the stereo pair, the relative positioning of which is adjusted with the Phase and Offset parameters. The envelope follower has optional sidechain input – see *Sidechain routing* on p47.

AUTO PAN

A modulated panning effect that can be used in synced or free-running mode. The effect employs dual amplitude-modulating LFOs to create stereo movement, the Phase of which can be set to zero for mono tremolo or rhythmic 'gate' effects.

BEAT REPEAT

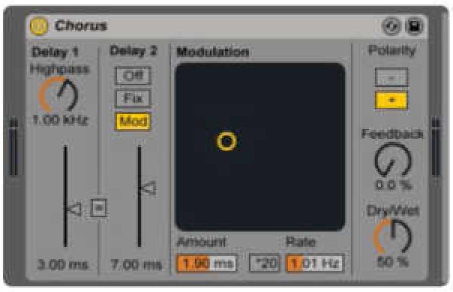
This buffer-based device generates stuttered repeats of the input signal, which can be filtered and pitched down to create glitch effects. A random factor is applied to determine whether Beat Repeat will repeat each 'slice' of audio or not, although this can be side-stepped by setting the Chance parameter to 100%. To mute the original signal when repeats are playing, switch the Output mode to Insert.

CABINET

A physically modelled guitar cabinet emulation, Cabinet is primarily intended to be placed after Live's Amp effect. You can emulate the effect of multiple mics on a cabinet by creating several chains in an Audio Effect Rack with a Cabinet effect in each, then using different Cabinet Microphone and Microphone Type settings for each chain.

CHORUS

Chorus is a classic modulation effect, and Live's straightforward take on it uses two



time-modulated delays to thicken sounds up. Chorus is typically used to widen instruments and vocals, but with subtler settings it can be used to tweak other sounds, too. For example, try it on tough mono percussion for a smoother stereo sound.

COMPRESSOR

Live's standard compressor is a transparent-sounding dynamics processor that's generally used to increase the average level of a signal or enhance its transients. The Peak, RMS and Expand modes determine how it responds to the incoming signal, while the sidechain input can be used to duck the signal when a specific, simultaneously input sound plays.

CORPUS

Bundled with the Collision instrument – see p33 – Corpus is an unusual device that uses physical modelling to simulate the acoustic characteristics of seven types of resonant object. The effect features a sidechain input for dynamically controlling the frequency of the resonance, which is otherwise set by the Tune parameter.

DYNAMIC TUBE

Simulates tube amplification, with three virtual tube models. Tube A won't produce distortions if the Bias is set low, becoming active only when the input signal is loud enough. Tube C produces distortion constantly, and Tube B sits between the two extremes of A and C.

Audio Effects

EQ EIGHT

Live's main equaliser is an eight-band parametric affair with stereo, left/right and mid/side modes, a choice of 12 or 48dB/octave low-pass and high-pass filters, and an integrated spectral analyser. Access the expanded view by clicking the triangular Toggle Display Location button at the top of the interface.

EQ THREE

This simplified three-band equaliser is primarily designed for live use, although it does have a few useful niceties for studio work. The high and low crossover frequencies are adjusted with the FreqLo and FreqHi knobs, the filter slopes toggle between 24 and 48dB/octave, and the kill switches below each band can be used to quickly deactivate and reactivate them.

EROSION

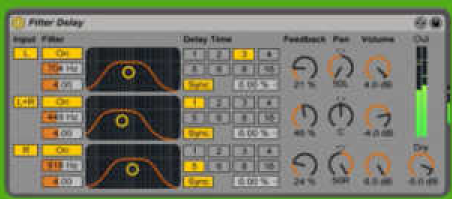
An unusual lo-fi effect, Erosion degrades the input signal by modulating a short delay with filtered noise or a sine wave, resulting in a unique, digital-sounding distortion. For enhanced stereo width, try Wide Noise mode, which uses independent noise generators for the left and right channels.

EXTERNAL AUDIO EFFECT

If you own any hardware effects, External Audio Effect makes integrating them into your Live sessions a breeze. Simply add the External Audio Effect device to a track and select the audio interface inputs and outputs to which your gear is connected. You can even enter a Hardware Latency value to ensure everything remains perfectly in sync.

FILTER DELAY

Live's most involved delay effect, Filter Delay features three independent delay lines with low-pass and high-pass filters. The delays take their input from the left, right and mid channels, and output to the same positions unless their Pan controls are adjusted.



FLANGER

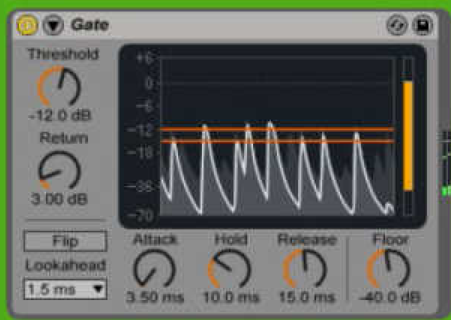
A classic psychedelic modulation effect, Flanger works much like Chorus in that it's based on two parallel, time-modulated delays. However, the delays are much shorter, delivering comb filtering effects. Try using it with a low Feedback setting and synced Rate to make mono percussion loops stereo in a fairly transparent manner.

FREQUENCY SHIFTER

Frequency-shifting works like pitchshifting, but moving each frequency by a fixed amount and thus not maintaining the harmonic relationships between them. This can create some truly strange timbres, particularly when a little of the original signal is mixed back in. Live's Frequency Shifter also includes a ring modulation mode, which modulates the incoming signal's amplitude level – great for robotic vocal effects!

GATE

When the input signal drops below a defined threshold, the Gate silences it. This useful for getting rid of background noise on audio tracks when nothing is playing, as well as tightening parts up. The Gate can be used creatively, too, by feeding its sidechain input a rhythmic source signal – known as "gating" or the "trance gate" effect.

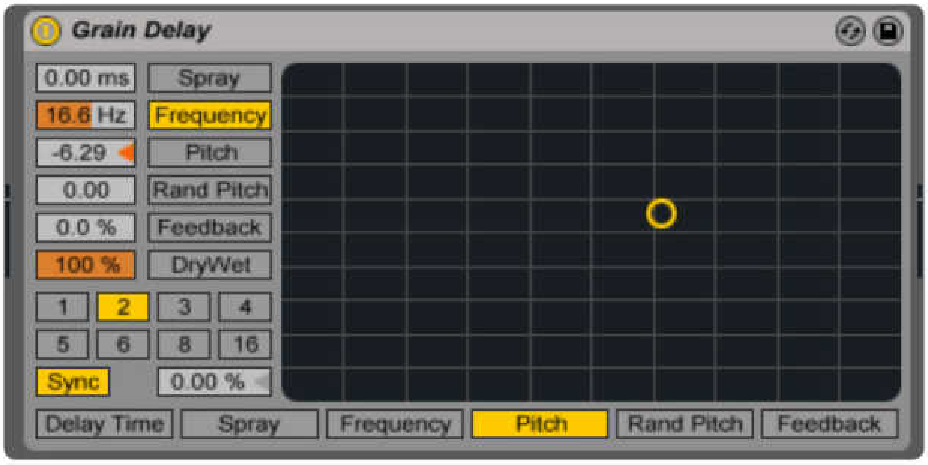


GLUE COMPRESSOR

A great alternative to Live's standard Compressor, Glue Compressor isn't as transparent, imbuing sounds with analogue character. It's perfect for beefing up beats, pumping up vocals or just adding a little warmth and grit to other instrument parts. The sidechain input makes it ideal for ducking effects, too. Glue Compressor is essentially Cytomic's acclaimed The Glue plugin, baked into Live.

GRAIN DELAY

Slices the incoming signal into tiny sections ('grains') that are individually delayed and repitched. Delay times and pitches can be randomised for abstract effects, and the big X/Y pad makes it the perfect tool for quickly creating twisted vocals and abstract FX.

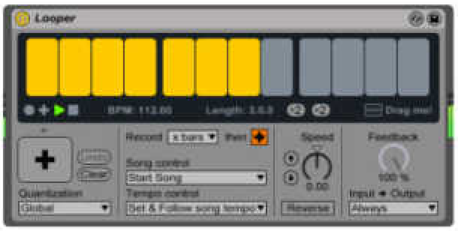


LIMITER

A mixing staple, Limiter prevents audio from exceeding a defined volume Ceiling – useful for taming signals that vary dramatically in volume level. The signal is compressed harder when shorter lookahead times are used, although this can result in increased distortion.

LOOPER

Based on hardware looping devices, Looper is a performance tool that continuously records and loops the input signal, overdubbing layer upon layer of sound onto a single track. Playback can be repitched and even reversed, making it up to the job of creating all manner of dramatic FX in real time.



MULTIBAND DYNAMICS

Live's most complicated dynamics processor, Multiband Dynamics splits the signal into three bands, each of which is capable of downwards and upwards compression and expansion. The effect offers a choice of peak or RMS modes, the ability to adjust crossover frequencies, and optional sidechain input.

OVERDRIVE

A guitar-style distortion effect, Overdrive makes tone-shaping easy, with its handy X/Y pad controlling a pre-distortion band-pass filter. Further timbral tweaking can be achieved with the Drive and Tone knobs, and the Dry/Wet mix knob opens Overdrive up to parallel processing, even when deployed as an insert effect.

PHASER

Another staple modulation effect, Phaser is a little different to Chorus and Flanger in that it uses a series of all-pass filters rather than delayed signals to generate its glorious sweeping sounds. Like Flanger, it can be used at low Feedback settings for subtle stereo widening effects.

PING PONG DELAY

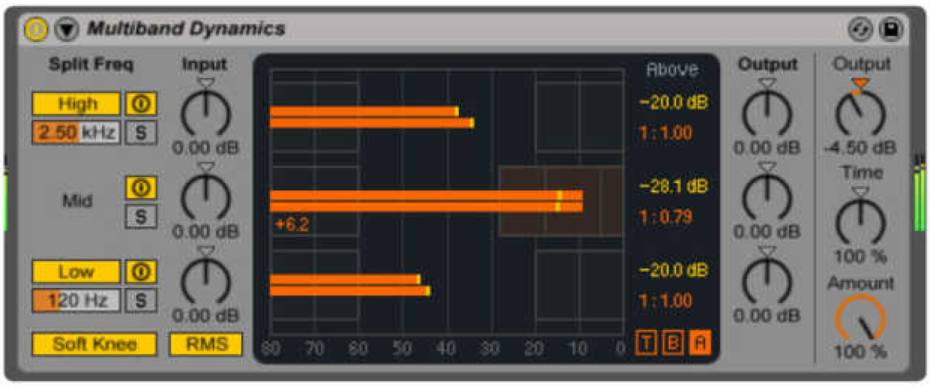
A quick option for adding texture, rhythm and stereo movement to tracks, Ping Pong Delay bounces a single delay line between the left and right channels. It's easy to set up: select a Beat Division to control the tempo of the delay, then adjust the wet (delayed) signal's band-pass filter with the X/Y pad to thin it out.

REDUX

This lo-fi effect doesn't have any frills, but it's still incredibly powerful. The Bit Reduction knob lowers the incoming signal's bit depth, while Downsample lowers the sample rate. These two processes work well together to recreate the sound of old-school hardware samplers.

RESONATORS

This one uses five parallel resonators to superimpose a particular tonal character on the source. Each resonator can be tuned in semitones, making it possible to add a complex musical quality to noisier sounds like beats or white noise sweeps.



Audio Effects

REVERB

Live's stock reverb effect is a fairly straightforward algorithmic affair with low-pass and high-pass input filters, high and low diffusion network filters, and a choice of Quality settings. Your average modern computer won't be taxed much by Reverb, so we suggest switching the Quality mode to High right off the bat.

SATURATOR

It might not look like much, but Saturator is enormously useful. Applying small amounts of Drive can manage a signal's peaks and optimise loudness, with more extreme settings creating obvious distortion effects. It also boasts a powerful Waveshaper - see *Saturator's custom Waveshaper* on p136.

SIMPLE DELAY

Simple but effective, this stereo delay provides separate left and right channels with independent (but linkable) Delay times that can be synced to the host tempo or set in milliseconds. Set the Dry/Wet level to 100%, then turn the Delay time of one channel down to 1ms and the other to under 20ms for a quick stereo widening effect.

SPECTRUM

A spectral analyser with automatic Y-axis scaling, Spectrum is ideal for visualising the frequency content of a signal. For more detailed examination, you can enlarge it by clicking the Toggle Display Location button, which pops the Spectrum Display out into its own panel. This is particularly handy for identifying the frequencies of specific partials.

UTILITY

One of Live's most useful effects, Utility can be used to mute channels, filter out DC offset, adjust gain, swap the left and right channels, use only the left or right channel, adjust pan position, control width, and invert the polarity of the left and right channels independently. When you want to automate a channel's volume level or pan position, using Utility, rather than automating the track directly, means you can still offset the track's parameters easily during the mixdown stage.

VINYL DISTORTION

Longing for the familiar grunge and grit of a battered LP or 12"? Vinyl Distortion creates even harmonic distortion and adds crackles to the source signal. It even comes with adjustable intensity, making it easy to get the perfect retro effect.

VOCODER

Live's Vocoder device is an absolute stunner. To create the classic robo-voice effect, put it on an audio track with a vocal on it, set the Carrier input to External, and pick a track with a synth on it in the Audio From panel. Try automating the Bandwidth and Formant controls for bizarre morphing effects, and activate the Enhance button to get a brighter sound.

Instruments

Live's selection of integrated instruments covers everything from sampling and virtual analogue synthesis to more esoteric fare such as physically modelled mallets, strings and electric pianos

MULTIBAND DYNAMICS

Designed to emulate the big sound and hands-on programming style of classic hardware synths, Analog might appear intimidating to the newcomer but it's actually quite straightforward once you get your head round the basics. The interface crams a lot of elements into a small space, but if you take the time to explore it fully, you'll find that there's actually nothing too complex going on.

Analog's raw sound is generated by two oscillators – each of which can be set to sine, saw, rectangle (pulse) or white noise shapes – and a dedicated noise generator. All three of these have Balance parameters, which control how their outputs are balanced between the synth's two resonant multimode filters. There are also dedicated envelopes for modulating pitch, filter cutoff frequency and amplitude, plus two LFOs for rhythmic modulation.

All pretty standard stuff, then, but Analog has a few tricks up its virtual sleeve, including sub-oscillators, oscillator hard sync, a choice of filter saturation models, and two- or four-voice unison detune with a Delay parameter that controls the lag between each unison voice. These bells and whistles give Analog the ability to produce some interesting and useful sounds, making it worthy of investigation even if you've already got a favourite subtractive synth. Analog is included with Live Suite, and costs £59 for Standard and Intro users.

COLLISION

A physical modelling instrument primarily dedicated to the synthesis of mallet percussion, Collision is capable of creating a wide palette of tones, from plucked strings to pan pipes. The synth's sound is produced by a pair of oscillators called Mallet and Noise, which model the behaviour of a mallet striking a surface and generate the instrument's initial impulse. This impulse is only a small component of the overall sound, though – it's the resonators that the impulse feeds that give the synth its characteristic timbre. There are two of them, each of which can be set to one of seven object types: Beam, Marimba, String, Membrane, Plate, Pipe and Tube. The characteristics of the object are customised via a selection of specialised parameters, such as Material (controlling the damping properties of the material used) and Inharmonics (governing the harmonic relationship of the partials generated by the resonator).

Collision makes full-bodied sounds that really cut through the mix, and by combining different resonator types, all sorts of weird, wonderful and atmospheric noises can be conjured up.

The instrument is included with Live Suite, is £59 for Standard and Intro users, and includes the Corpus effect (see p29).

DRUM RACK

Live's Drum Rack is a container device used for building drum kits. A Drum Rack is automatically created when you use Live's slicing functionality – see *Audio slicing and creating custom Library content* on p19 – but their main purpose is to enable you to design your own kits.

Each of a Drum Rack's slots/pads is triggered by a specific MIDI note, from C2 to G8, and hosts its own full device chain, so you can drop whatever instruments or effects you like on each one – Simpler is often the first port of call for sampled hits, thanks to its ease of use and low CPU overhead.

The potential is there to creating massive, complex kits, but the container is surprisingly light and quick to use, making it ideal for building more basic things such as Choke Grouped hi-hats, too. To do exactly that, activate the Show/Hide Chain List and Show/Hide Input/Output Section buttons, and set both pads to the same Choke Group. Now, whenever you play one sound on the Choke Group, the other will immediately stop sounding.

ELECTRIC

Electric emulates the sound of classic 70s electric pianos. Like Collision, the sound starts with synthesised Mallet and Noise elements, which 'strike' a physically modelled fork. The fork is divided into two subsections: Tine (the part of the fork struck directly by the mallet) and Tone (controlling the secondary resonance of the fork), and by adjusting their parameters, a variety of timbres can be generated. Electric also simulates the dampers found in real electric pianos with its Damper section, which gives the release stage its character, and magnetic coil pickups with its Pickup section, which affects how bright and distorted the sound is.

Electric does a great job of creating convincing electric piano tones, which can

Instruments

be enhanced through the application of Auto Filter, Phaser, Amp, Cabinet, Reverb and Auto Pan (for tremolo) from Live's Audio Effects. Check out the EP Faceplant preset for an example of how these effects can be used in conjunction with Electric in the Instrument Effects rack to create a classic vintage sound. Electric is included with Live Suite, and is £59 for Standard and Intro users.

EXTERNAL INSTRUMENT

External Instrument makes integrating MIDI hardware instruments into your Live sessions a piece of cake. With your hardware instrument hooked up to your Mac or PC's audio and MIDI interfaces using the usual cables, add External Instrument to a MIDI track, select the port you want to send MIDI data from and the audio input the hardware is connected to, and you're ready to rock! Live even adjusts for latency, making it a truly viable way to make your old-school kit useful again.

IMPULSE

A drum sampler with complex modulation capabilities, Impulse is a quick and fun alternative to the considerably more powerful (but very different) Drum Rack. It features eight slots onto which samples can be dragged from the Browser or the arrangement, which are triggered on consecutive white notes from C3 up. Each sample slot has its own individual set of parameters, many of which can be modulated via velocity or randomised.

The parameters are divided into four sections, the first of which contains the Start, Transpose and Stretch controls. Using these, you can adjust the start point of the sample, change its pitch, and even timestretch it with a choice of two algorithms.

The second section is a basic saturator with on/off and Drive controls. The third section, a resonant filter, features low-pass, high-pass and band-pass modes, each with a choice of slopes, plus a notch mode.

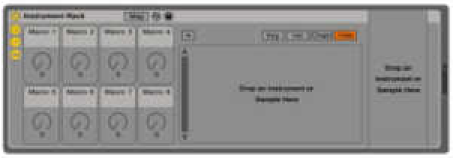
Finally, the Pan and Volume section includes those eponymous controls plus a Decay time setting with two modes: Trigger has the sample beginning to decay with the note-on message, while Gate starts to decay upon receiving a note-off message. At the right-hand side of the fourth panel are global controls for Volume, Time (controlling both Decay and Stretch) and Transpose.

INSTRUMENT RACK

This endlessly useful container device enables composite instruments to be constructed by layering up parallel device chains. For example, you could build your own 'hybrid' synth, with an Operator instrument providing the sub-bass and an instance of Analog generating a wider mid sound on top of it, both triggered by the same MIDI part.

Audio effects also work in Instrument Racks, and you can use VST/AU plugins, too.

Key and velocity splits can be set up for each chain, and you can control which chains are playing directly using the blue Chain range bar and Chain Selector parameter. Like the Effect Rack, the Instrument Rack features



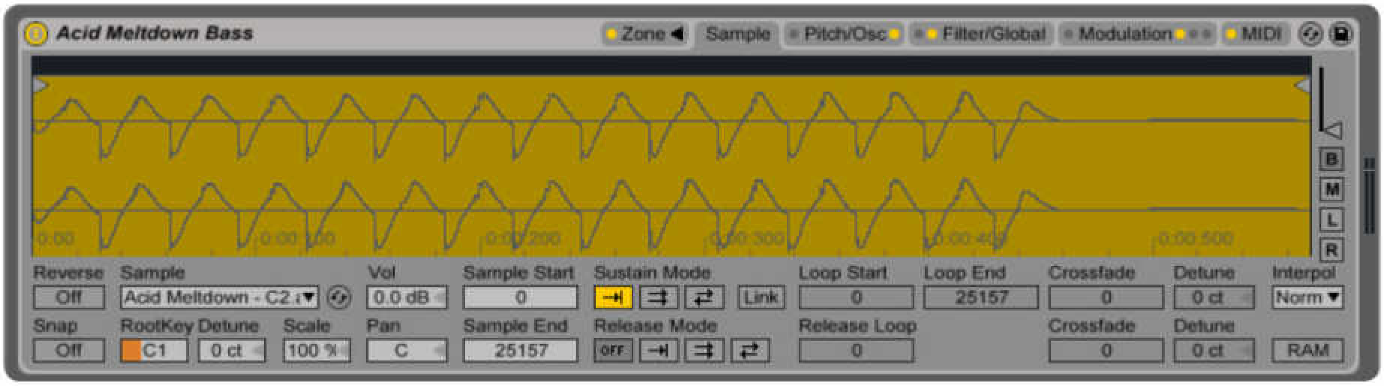
eight Macro knobs, each of which can be freely assigned to multiple parameters.

OPERATOR

Live's fabulous frequency modulation (FM) synth might not have the specs of its third-party plugin rivals, but it's well designed and can produce some strong sounds, especially in the bass and lead departments.

FM synthesis creates particularly characteristic timbres by modulating the pitch of one oscillator (known in FM jargon as an 'operator') with the amplitude of another.

While most modern soft synths have a modulation matrix that allows the user to set up new routings on the fly, Operator's design is a throwback to a breakthrough hardware FM synth, the Yamaha DX7. The fixed approach employed by the DX7 and Operator presents a choice of fixed 'algorithms' that determine the modulation routing, and while



it's undoubtedly a less flexible system than a modulation matrix, it does make the synth less intimidating for novices. Operator also has just four operators compared to the six used by most FM soft synths.

Despite its relatively simple design, Operator still isn't quite as intuitive as a virtual analog synth, so to learn more about programming it check out *Getting to grips with Operator* on p38, *Smooth Operator* on p94, and *Creating a waveshape with Operator's additive synthesis* on p141. Operator is included with Live Suite, and is £59 for Standard and Intro users.

SAMPLER

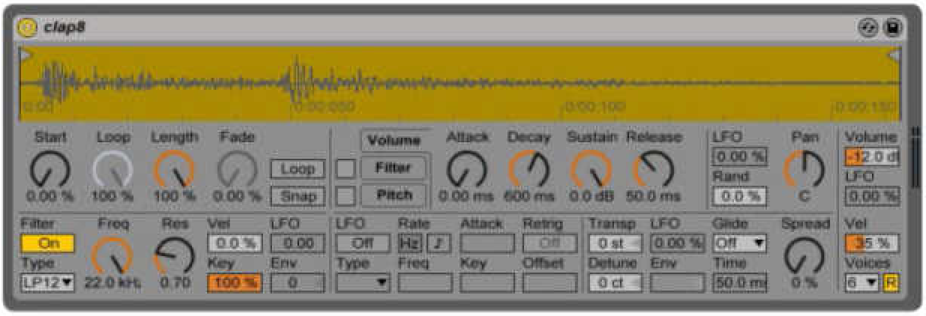
While Simpler makes light work of basic sample playback and triggering, Sampler offers many more features, including ping-pong looping, reverse mode, flexible envelopes, a choice of filter modes, a waveshaper, adjustable pitchbend range, more modulation sources, and the ability to load multiple samples and split them into key and velocity zones. These capabilities make it a powerful tool for sound design, and because you can drag and drop clips directly into it from Live's audio tracks, it also makes resampling incredibly quick and painless. To learn more about Sampler, check out *Sound design with Sampler* on p40.

Sampler is included with Live Suite, and costs £59 for Standard and Intro users.

SIMPLER

This basic, no-frills sampler is a fundamental component of Live, and in fact, you might have even used it without realising it, as it's used for playback of sliced clips generated by Live's Slice to New MIDI Track function (see p20) as part of an Instrument Rack, and is only revealed when the Rack's Show/Hide Devices button is clicked. Live will also automatically create a Simpler instrument when you drop an audio file onto a MIDI track – this is a great way to quickly turn samples into playable instruments.

Simpler enables a single sample to be played chromatically up and down the keyboard, repitching it as it does so – ie, speeding the sample up for higher notes and slowing it down for lower ones. A resonant multimode filter is provided, as are looping capability with crossfade; dedicated volume, filter and pitch envelopes; an LFO; a choice of



portamento and glide modes; and the ability to 'spread' samples, doubling and detuning them for stereo width.

The R button at the bottom right-hand corner of the interface tells Simpler to retrigger voices rather than creating an additional voice when a note that's already playing is played again – handy for patches with long release times, when you want a more defined sound.

TENSION

Live's dedicated string synth, Tension generates its sounds by physically modelling one of four types of 'Excitator' (Bow, Hammer, Bouncing Hammer, and Plectrum), a Damper (which reduces the decay time of the strings' vibration), the String itself, 'Termination' (modelling the interaction between fret, finger and string), a Body (Piano, Guitar, Violin or Generic), and a Pickup mic with variable positioning. The instrument also has a secondary global settings and resonant filter page, the latter featuring ten modes, including low-pass, high-pass, band-pass, notch and formant, as well as dedicated envelope and LFO modulation sources.

Unfortunately Tension isn't the most intuitive of synths, involving a fair bit of trial and error to get to grips with. A good way to find out what it's capable of is to check out the included presets, which will give you some clues as to how it can be employed to create string, guitar and mallet sounds. The presets also include some more abstract patches, such as pads, evolving ambient sounds and FX, which use extreme settings to create rather severe noises of the kind that you'll either love or hate.

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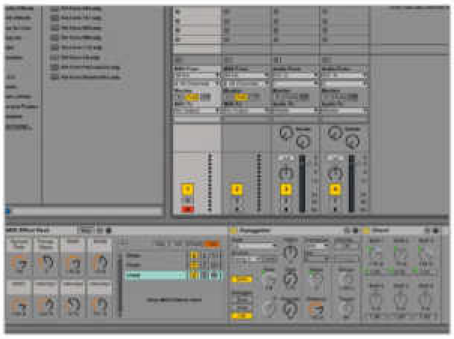
> Step by step

15. A whistlestop tour of Live's MIDI Effects

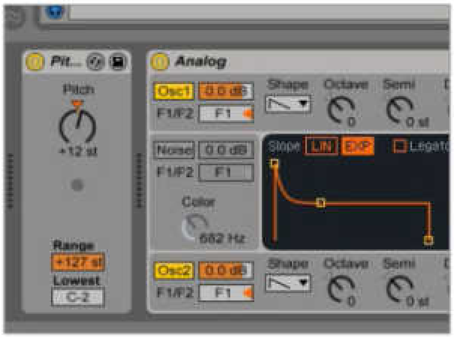
1 > Live's MIDI Effects open up a world of new sequencing and performance potential. Let's start our run-down with **Arpeggiator.** This powerful arp can take chordal input and turn it into monophonic sequences in any of 18 styles. The most important parameters are **Steps**, which controls how many octaves the sequence plays on, and **Rate**, which determines how fast the sequence plays.

2 > **Chord:** Simple but useful, Chord adds up to six transposed versions of every note it receives. For each 'voice' you can choose a transposition amount (up to +/-36 semitones), and a relative velocity (between 0 and 2). Chord is great for adding octaves and fifths to parts, and generating ravey chord sequences.

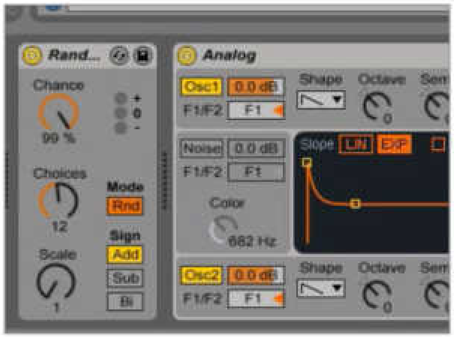


3 > **MIDI Effect Rack:** Just like the Audio Effect and Instrument Racks, the MIDI Effect Rack enables you to create chains of MIDI Effect devices. The only limit is your imagination! Oh, and the maximum polyphony of the instrument being triggered by it, of course. Like the other Racks, the MIDI Effect Rack has its own macros, gearing it up nicely for live use.

4 > **Note Length:** The straightforward name bestowed upon this one leaves no ambiguity as to what it does, which is transform the lengths of the notes pushed through it. In **Time** mode, it changes all notes to a specific duration in milliseconds, while **Sync** mode lets you select from 64th-note to quarter-note resolutions. It can also be triggered by MIDI note-on or note-off messages.



5 > **Pitch:** Despite being the simplest of all the MIDI Effects, Pitch is probably the one you'll use the most, as a super-convenient way to transpose MIDI tracks. Pitch lets you transpose the note input by up to +/-128 semitones, with the **Lowest** and **Range** parameters used to filter out notes above or below specific pitches.



6 > **Random:** Why compose your own music when you can leave it to the Fates? Random transposes the incoming note pitch within the range defined by the **Choices** knob, which can be multiplied by the **Scale** parameter. This transposition mode can be set to either add or subtract, and for the curious there's a 'bi' mode, which does both.

7 > **Scale:** This cheeky effect automatically remaps the pitch of the note input, snapping it to one of a variety of scales – a whopping 59 of them, to be precise, including everything from majors and minors to numerous modes (Locrian, Mixolydian, etc) and more. You can edit these and even create your own scales using the intuitive Note Matrix.

8 > **Velocity:** Another hugely useful effect, this one transforms the velocity of incoming notes, essentially working like a MIDI gain control. Naturally, it's particularly convenient when working with instruments that change timbre over their velocity range, enabling you to turn delicate MIDI clips into thunderous ones and vice versa.

POWER TIP

>All together now

Each of Live's MIDI Effects is powerful on its own, but the sparks really start to fly when they're used in conjunction with each other. For example, try putting a Chord with multiple voices active in front of an Arpeggiator set to a few **Steps** and a fast **Rate** for insane melodic synth riffs playable with just a single finger – ideal for those chiptune moments! Don't forget that MIDI Effect parameters can be automated, so you can easily modulate MIDI parts throughout your composition.

Getting to grips with Operator

Frequency modulation (FM) synthesis has a reputation for being difficult, but it doesn't really deserve it. Although many FM synths are rather involved, the technique itself isn't particularly complicated or time-consuming once you've got your head around it. Operator is by no means the most convoluted of instruments, but its quirky design (inspired by Yamaha's classic DX7 hardware synth) means it's tricky to figure out for yourself if you're unfamiliar with the principles behind FM.

Most synths you'll have come across (including Live's Analog) are based on subtractive synthesis, meaning that their sound comes from oscillators generating harmonically rich waveforms, like sawtooth and square shapes. These are then sculpted with filters, which attenuate specific frequency ranges. FM works the other way around, typically starting with a simple sine waveform that only has a single harmonic. FM oscillators are often referred to as 'operators' (hence Operator's name, though Ableton confusingly refer to Operator's operators as oscillators!), and more complex harmonics are generated by modulating a voiced (heard) operator's pitch with another (often unvoiced) operator's amplitude. The effect that this has depends on the frequency of the modulating operator: at low frequencies, you'll only hear pitch modulation (ie, vibrato), but as the frequency of the modulating operator increases, sideband partials begin to appear above and below the fundamental frequency in the voiced operator's output. So, the positioning of these partials (and therefore the voiced operator's timbre) can be affected by changing the pitch of the modulating operator. The amplitude of the partials is determined by the level of the modulating operator, and by using an envelope to adjust it, we can create sounds that change in timbre over time – much like using an envelope to control the filter cutoff of a subtractive synth.

This may all seem impenetrable if you're unfamiliar with synthesiser jargon, but in the following walkthrough, you'll see that, actually, FM synthesis makes for a quick, effective and fun way to create sounds.

> Step by step

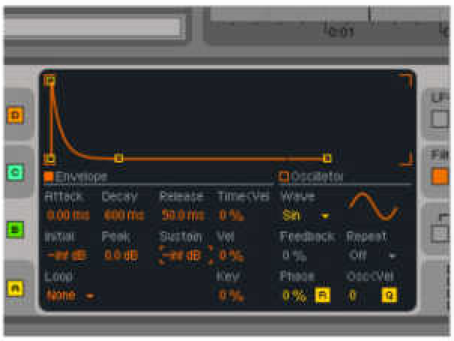
16. Basic frequency modulation synthesis with Operator

1 > Drag Operator from Live's Instrument folder onto a MIDI track and play a few notes. You'll hear the synth's default sine tone – it's very simple, and if you drag Spectrum from the Audio Effects folder onto the track, you'll see that it only has a single harmonic. (Audio: **Sine tone.wav**)

2 > Operator's four oscillators (A-D) are found on the left-hand side of the synth, and to the bottom right of the central panel is a set of coloured boxes. These represent Operator's oscillators, and the order in which they're arranged shows how the synth's routing is currently set up. Any oscillators on the bottom row (only A in this case) are voiced, and the rest are unvoiced.

3 > If an oscillator is positioned above another oscillator, it means it modulates that oscillator, so here, A is modulated by B, B by C, and C by D. If an oscillator doesn't have another oscillator above it, it can modulate itself; so in the configuration shown above, Oscillator D is self-modulating. Let's use Oscillator B to modulate Oscillator A: turn B's **Level** knob up and the sound will begin to change. (Audio: **Simple FM.wav**)

4 > With Oscillator B's level turned all the way up to **0dB**, we get something approaching a sawtooth wave. Next, turn up Oscillator B's **Coarse** level. The higher it goes, the harsher the timbre becomes. This doesn't sound enormously pleasant when it's sustained, so let's modulate Oscillator B's level to make the patch more dynamic. Set Oscillator B's **Coarse** knob to **4**. (Audio: **Pitched modulator.wav**)



5 > As we've been editing Oscillator B, its envelope will already be displayed in the large panel to the right. Drag the envelope's **Sustain** down to **-inf dB**. Now you'll hear that the modulation level (and therefore the level of harmonics in the signal) starts high and quickly drops, giving us a punchy, techy bass sound. (Audio: **Techy bass.wav**)

6 > We can refine this sound by lowering Oscillator B's level. Set it to **-9dB** and you've got yourself a deep house-style bass. To mellow it out further, turn Oscillator B's **Coarse** down to **2**. That's basic FM synthesis in a nutshell: modulating one oscillator with another, and adjusting the modulating oscillator's amplitude envelope and pitch until you get the sound you're after. (Audio: **Deep house bass.wav**)

7 > Let's look at some more advanced techniques. Set Oscillator C's Level to **-9.0dB**. Now Oscillator C is modulating B, which modulates A. This patch gives us a simple plucked guitar-style sound. Turn Oscillator C's **Coarse** up to **7** for a clavichord. Clearly, FM is an extremely flexible synthesis technique, capable of creating all kinds of synthetic and 'acoustic' sounds. (Audio: **Syn clav.wav**)

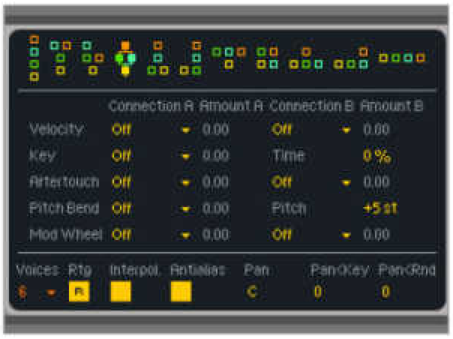
8 > Set Oscillator D's level to **0.0dB** and turn its **Coarse** up to **2**. This makes the timbre harsher, but we can go even further by using the oscillator self-modulation capabilities mentioned earlier, which are controlled by the **Feedback** parameter in the central panel. Turn it up to **100%** to get a very harmonically rich sound. (Audio: **Rich harmonics.wav**)

9 > So far we've had our oscillators tuned relative to the MIDI note input, but they can be set to fixed frequencies, too. Click Oscillator D's **Fixed** button to activate it. The **Coarse** button changes to **Freq**, and we can now set a Hz value for the oscillator. Play the patch back and increase the **Freq** value. You can create some cool FX by automating **Freq**. (Audio: **Automated Freq.wav**)

10 > This is all getting a bit noisy, so deactivate Oscillator D. On the right-hand side of Operator's interface are various parameters that we can use to tweak the sound created by the oscillators on the left. For example, we can control the envelope times of all the oscillators at once with the **Time** knob. Turn it up to 100% for a more sustained sound. (Audio: **Long clav.wav**)

11 > Spread is another useful parameter. So far, all the sounds we've generated with the synth have been mono, but activating **Spread** causes Operator to use two panned and detuned voices per note, for variable stereo width. Turn **Spread** up to **100%** to make the sound as wide as possible. (Audio: **Stereo clav.wav**)

12 > 'Pure' FM synthesis means using only sine-based oscillators to make sounds, but Operator can use other wave shapes as well. Click Oscillator C's panel to select it, and in the central panel, click the **Wave** parameter and select **Square D** from the list. The waveform shape to the right will change to a square, and as a result, the timbre will change radically. (Audio: **Square modulator.wav**)



13 > Click the oscillator arrangement at the bottom right-hand corner of the main panel and you'll see the 11 predefined oscillator arrangements, or 'algorithms', at the top of the main panel. Click the fourth one in from the left to select it. This heavily alters the sound, because Oscillator C is now modulating Oscillator A directly, rather than via Oscillator B. (Audio: **Alternative algorithm.wav**)

14 > Another way in which Operator deviates from pure FM synthesis is that it features a filter with a built-in waveshaper. Click the **Filter** panel on the right-hand side of the interface, then, in the central panel, set the **Shaper** mode to **4Bit** and turn the **Drive** up to **9.0dB**. That should grunge up the patch nicely. (Audio: **Lofi synth.wav**)

POWER TIP

>Going loopy

Operator features an LFO hardwired to the pitch/frequency of each oscillator and the filter's cutoff frequency. It can also be assigned to a secondary destination, including the amplitude level of any one of the four oscillators. However, LFO-style effects can be applied to all four oscillators' amplitude levels via the envelope's **Loop** parameter, which can be set to four modes: **Loop** (the envelope retriggers at the end of the decay stage), **Beat** (the envelope loops, unquantised, at a division of Live's project tempo), **Sync** (like **Beat** mode but quantised) and **Trigger** (ignores note-off messages – useful for playing drum sounds live).

Sound design with Sampler

Live has a great built-in sampler in the shape of Simpler – a straightforward, lightweight instrument that's ideal for basic playback of single samples. It includes a few creative tools such as looping, a multimode resonant filter and dedicated envelopes for filter cutoff, pitch and volume, but it's pretty limited in many areas. For example, you can't even change its pitchbend range, which is fixed at 5 semitones.

Thankfully, Simpler's big brother, Sampler (£59 or included with Live Suite), is absolutely stuffed with awesome sound-mangling tools. In fact, we'd go as far as saying it's one of the best software samplers on the market. Its most popular rival, Native Instruments' Kontakt, is absurdly capable, but Sampler is far more streamlined and immediate, and despite being less complex, it still has enough power and depth to happily serve as your go-to sampler. It's an instrument that really makes the creative possibilities of sampling come alive.

Sampler comes with a wealth of features that make it one of Live's most creative instruments

The reason samplers are so useful is that they allow you to take any sound – be it a 'found sound' recording, synthesised material or, yes, even bits of other people's music – and turn it into something new and inspiring. Sampler is the perfect instrument for creatively twisting and processing audio, thanks to its multiplicity of features, including morphing filters, a choice of looping modes, waveshaper, modulation oscillator, and the ability to create key and velocity splits for multisampling.

In the walkthrough below, we're going to show you the basics of creating a single-sample patch in Sampler, from loading and looping an audio file to using the filter and modulation oscillator. For more, check out *Key Zone splits with Sampler* on p137.

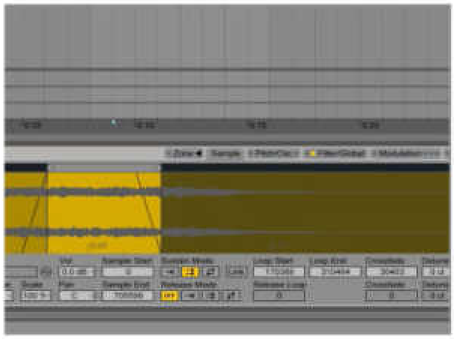
Before we get started, you might find it useful to know that any instance of Simpler can be converted into a Sampler by right-clicking its Device Title Bar and selecting Simpler»Sampler. It couldn't be, well, simpler!

> Step by step

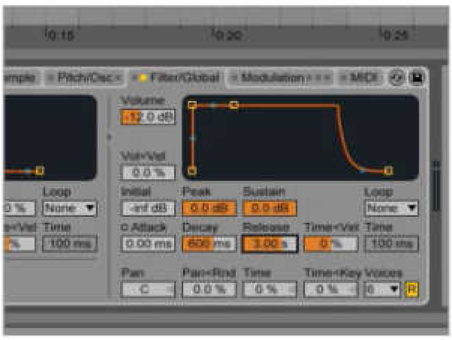
17. Creating a dynamic pad patch with Sampler

1 > Drag Sampler from the Instruments folder onto a MIDI track, then drag **Sweet pad.wav** onto the **Drop Sample Here** panel. We'll begin by looping the pad so that we can hold sustained notes with it indefinitely. In the **Sustain Mode** panel, click the second button from the left to switch to **Loop** mode.

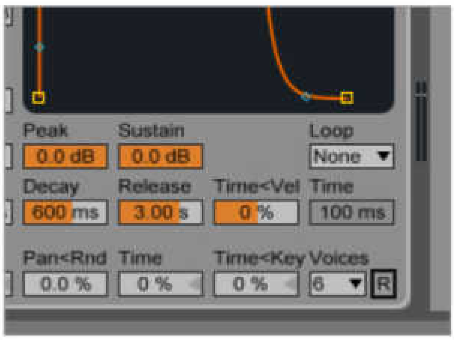
2 > We want the looped section to have a consistent sound and to feel natural. Drag the **Loop Start** to around **170000** and the **Loop End** to about **310000**. When you play the patch, you'll hear that this loop is already pretty smooth, but it's still possible to tell when the 'playhead' moves from the end of the loop to the start. (Audio: **Straight loop.wav**)



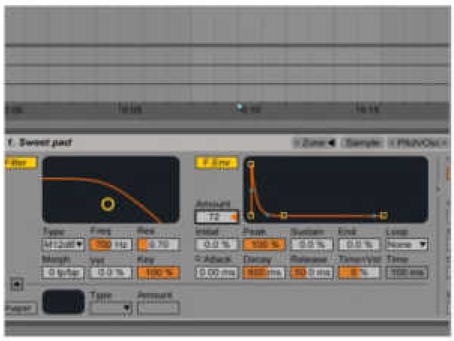
3 > We can make the looping less obvious using Sampler's **Crossfade** parameter. Turning it up to **30000** or so vastly improves the smoothness of the loop. The next thing we need to contend with is the fact that the patch stops dead when the instrument receives a note-off message. (Audio: **Crossfade loop.wav**)



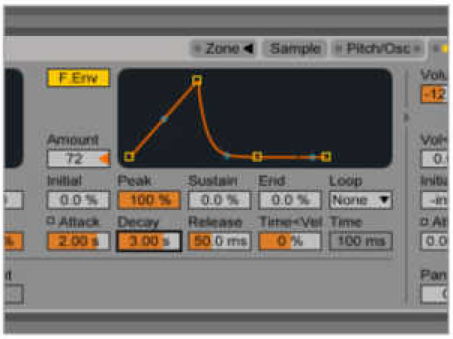
4 > Click the **Filter/Global** tab at the top of the instrument's interface. In the **Volume** panel you'll see that the **Release** time is set to **50.0ms**; set it to **3.00s**. Now, the sample should fade out as smoothly as it fades in. (Audio: **Lazy release.wav**)



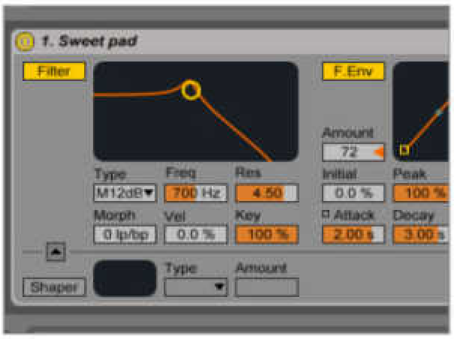
5 > If you play a note twice, with the second landing before the first's release stage has finished, the first note will end abruptly. This might be desirable with hi-hats, but it sounds unnatural with our pad sound. Click the **Retrigger (R)** button at the bottom right-hand side of the interface to deactivate it. We can now trigger new notes without killing the previous ones.



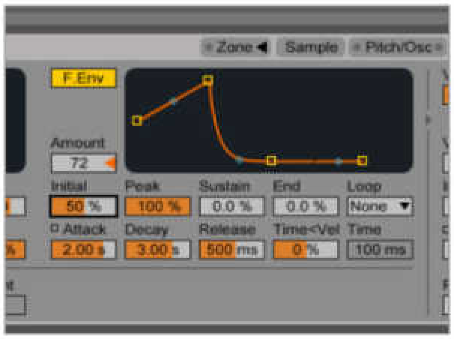
6 > So far, so good: we've got a basic pad patch that sounds natural. However, we're barely scratching the surface of Sampler's capabilities. Let's try something more creative. In the **Filter** panel, drag the **Freq** (filter cutoff frequency) fader down to **700Hz**, and in the **F. Env** (filter envelope) panel, set the **Amount** to **72**.



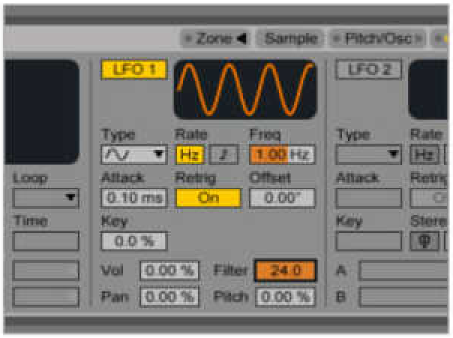
7 > The filter envelope is now modulating the cutoff frequency by the maximum possible amount. However, it's currently set to an instantaneous attack and relatively quick decay, and because the pad takes a while to fade in we can't really hear the filter movement. Set the filter envelope **Attack** to **2000ms** and **Decay** to **3000ms**. (Audio: **Filter sweep.wav**)



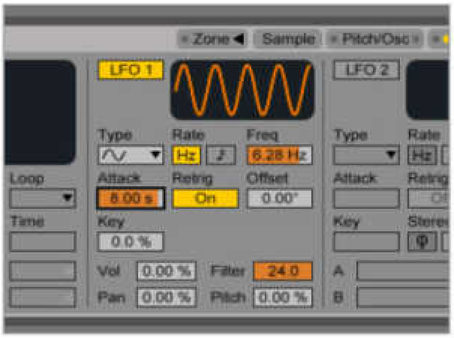
8 > To make the filter sweep more obvious, turn the **Res** (resonance) parameter in the **Filter** panel up to **4.50**. The patch is starting to get more character, and the filter envelope's short **Release** time gives us an interesting little accent on the end of each note as the filter cutoff frequency quickly drops to zero. (Audio: **Resonant filter sweep.wav**)



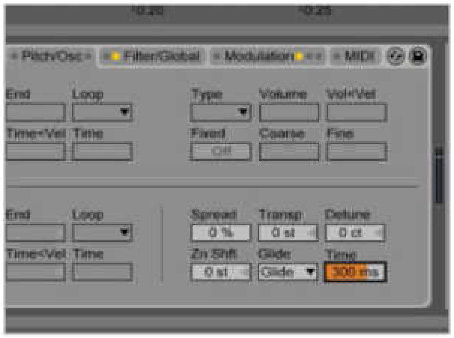
9 > This is a cool effect, but it's a bit much. Turning the filter envelope **Release** up to **500ms** dials in a good amount of character without going overboard. Another useful tweak is to turn the filter envelope's Initial level up to **50%**, causing the filter to open faster. (Audio: **Tweaked filter envelope.wav**)



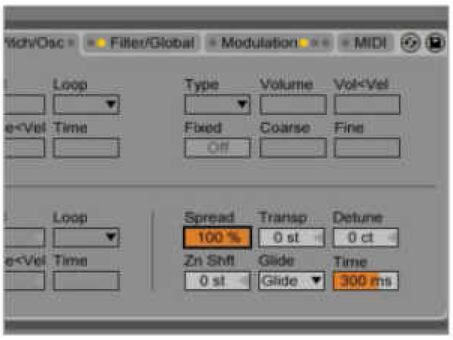
10 > As well as its dedicated envelope, we can also modulate the filter with Sampler's LFOs. Click the **Modulation** tab, then the **LFO1** button to activate it. At the bottom of the **LFO1** panel, you can see that it's hardwired to **Volume**, **Pan**, **Pitch** and **Filter**. Turn the **Filter** fader up to **24.0**. (Audio: **LFO modulation.wav**)



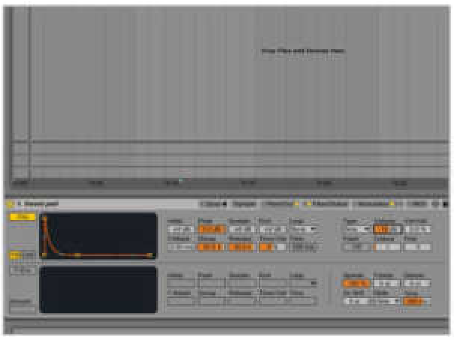
11 > The LFO movement is a little slow, so set its **Freq** to **6.28Hz**. This effect sounds pretty cool, but the flickery motion that's been introduced to the filter envelope's attack stage doesn't really add much to the sound. Turn the LFO's **Attack** up to **8.00s** to bring it in gradually, so that its effect is only obvious after the filter envelope reaches its sustain stage. (Audio: **Fast LFO.wav**)



12 > Gliding between notes can work well with pads, so let's try that out. Click the **Pitch/Osc** tab, and switch the **Glide** mode from **Off** to **Glide**. The default setting of **50ms** is too brisk, so turn it up to **300ms**. (Audio: **Slow glide.wav**)



13 > Like many of Live's instruments, Sampler has a **Spread** knob that uses two detuned, panned voices for each note in order to enhance stereo width. Turn **Spread** up to **100%** and bask in that ultra-wide sound. (Audio: **Wide pad.wav**)



14 > Sampler even has synth-style frequency and amplitude modulation capabilities. Click the **Osc** button to activate the modulation oscillator. This is set to **FM** mode by default, but it won't make any difference to the sound until you turn the oscillator's level up. Set the modulation oscillator's **Volume** fader to **-18dB** and you'll hear the eerie, robotic effect it has on the patch. (Audio: **Frequency modulation.wav**)

>Best practice

Sampler offers a choice of interpolation modes. Interpolation is what a sampler does when it transposes notes, and the algorithm used to do it will affect the character of all transposed notes. This difference is particularly noticeable on sounds with a lot of high-frequency content, such as cymbals. Select a mode from the **Interpol** menu in the **Sample** page. **Good** and **Best** modes place slightly higher demands on your computer's CPU, but it doesn't amount to much on a modern machine, and you'll likely find the improved sound worth the price.

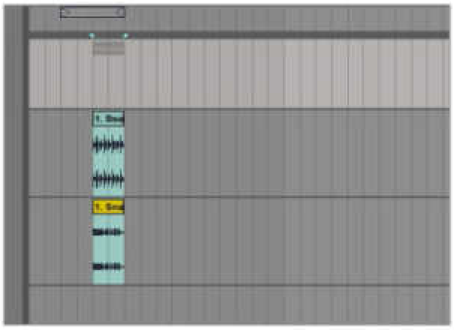
Finders keepers

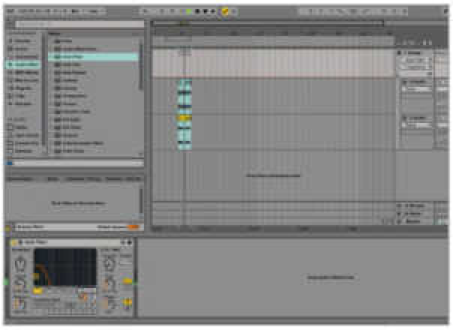
Live's Browser has a handy timesaving feature in the form of its search field, which can be accessed by pressing Ctrl/Cmd+F. Typing a word or phrase into the field searches the currently selected Browser category for anything containing that text, which can be useful for quickly locating specific instruments, audio effects and presets, as well as samples and clips. You can even search all categories together by selecting All Results at the bottom of the Category panel after you've entered your search term. It's also possible to search locations in the Places panel; and to search a particular folder on your system, add it with the Add Folder button at the bottom of the list.

Another helpful aspect of the Browser is that it includes a Current Project folder location in the Places panel. This allows you to see all the audio and MIDI clips in the currently active Live Project, and when you're working on larger projects it can often be quicker to locate things here rather than scouring the Arrangement view for them – especially when used in conjunction with the search field.

Access all the audio and MIDI clips contained in the track you're working on in the Current Project section

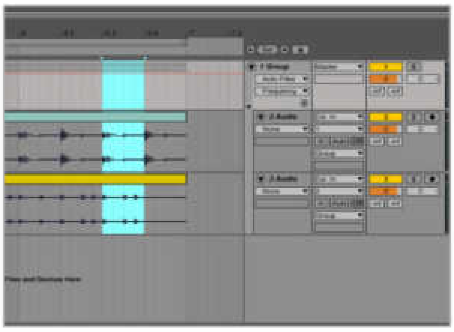
> Step by step 18. Working with grouped tracks

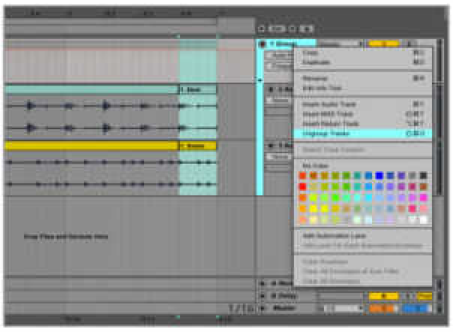




1 > Grouping multiple tracks enables us to edit and process them as a single entity. Drag **Beat.wav** and **Snare.wav** onto audio tracks in the Arrangement view. To select both tracks, click the first, hold **Shift**, then click the second. Press **Ctrl**/**Cmd**+**G** to group them.

2 > This creates a Group 'folder' track, and places the selected tracks inside it. We can now process both tracks as a single, grouped signal. Drag **Auto Filter** from the **Audio Effects** folder onto the Group track and turn the **Cutoff** down. On playback, you'll hear both sounds being filtered out.

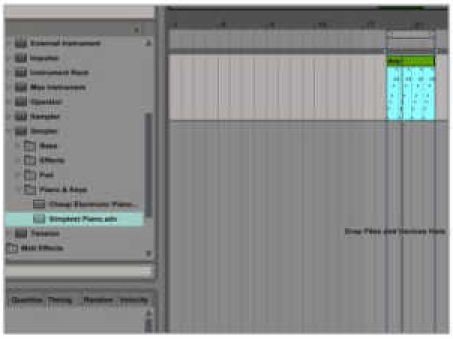


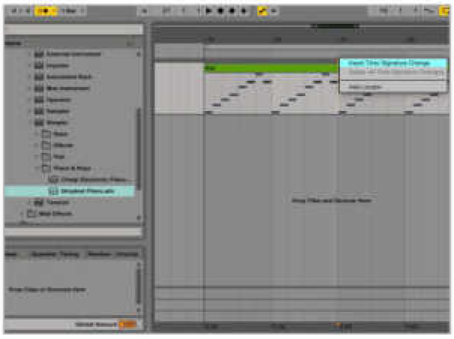


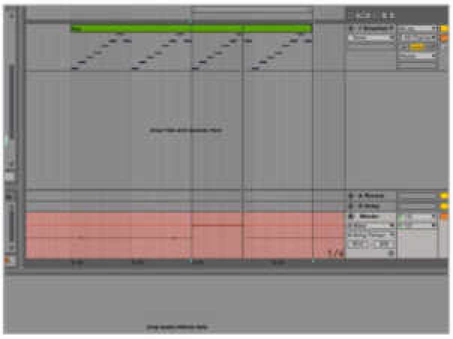
3 > Turn the filter **Cutoff** back up, then zoom in by dragging down on the Beat Time Ruler. Drag over the seventh beat on the Group track. This highlights the Group track and both Audio tracks, indicating that any edits we perform will affect everything in the Group. Press **Ctrl**/**Cmd**+**D** to duplicate the selected audio.

4 > This duplicates the audio material on both the Beat and Snare tracks. You can add more tracks to the Group by dragging them onto the Group track, and disband the Group by right-clicking it and selecting **Ungroup Tracks**.

> Step by step 19. Tempo and time signature changes







1 > Drag **Arp.mid** onto a MIDI track in the Arrangement view (you don't have to import tempo or time signature information when prompted), then open **Packs/Core Library/Devices/Instruments/Simpler/Piano & Keys** and drag **Simplest Piano** onto the MIDI track. Click the **Metronome** button to activate it, select the MIDI clip, press **Ctrl**/**Cmd**+**L** to loop around it, then press the **Play** button.

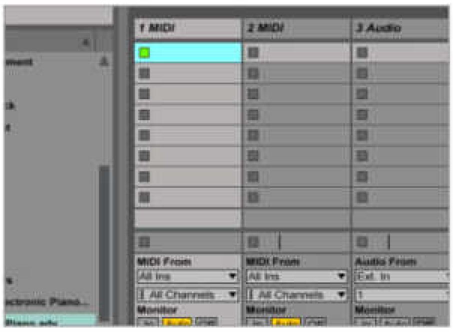
2 > Having the metronome active makes it possible to hear the effect that changes to the time signature have. The **Time Signature Numerator** sets the number of beats in a bar, and the **Denominator** determines the note value of those beats. You can automate time signature changes by right-clicking the Scrub Area and selecting **Insert Time Signature Change**.

3 > To automate the **Tempo**, right-click it and hit **Show Automation**. This sets the Master track's automation lane to the project tempo, which you can automate like any other parameter. The easiest way to create a tempo change for a specific region is to set it to loop, set the **Tempo** parameter to the desired value, click **Record**, then click **Stop** once the region has finished recording.

> Step by step 20. MIDI and key mapping

1 > A big part of what makes Ableton Live such a powerful tool for jamming and playing live is the fuss-free way in which you can set up MIDI controller and QWERTY keyboard mappings. In the Session view, click the **Key** button or press **Ctrl**/**Cmd**+**K** to enter **Key Map** mode.

2 > You'll have more keys to play with if you deactivate Live's Computer MIDI Keyboard – click the keyboard icon located to the left of the **Key** button to do this now. You can assign a key to anything coloured brown in **Key Map** mode. Click the clip at the top left-hand corner of the Session view, and press **Q** to assign the key to that clip.



3 > Click the **Key** button or press **Ctrl**/**Cmd**+**K** again to deactivate **Key Map** mode. Now, when you press the **Q** key you'll see the **Stop** button on the assigned clip light up. Note that all assignments are saved with Live Sets and reset when you create a new Set.

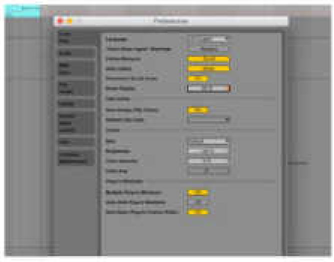
4 > To make hardware MIDI controller assignments, click the **MIDI** button to the right of the **Key** button, or press **Ctrl**/**Cmd**+**M**. Click the purple button or parameter you want to control, then move a control on your hardware to assign it. When you're done making assignments, click the **MIDI** button or press **Ctrl**/**Cmd**+**M** again to exit **MIDI Map** mode.

Screen time

Live features a couple of options for making the most of your computer's monitor or monitors, the most basic of which is Full Screen Mode. To activate this in OS X, click the green maximise button at the top left-hand corner of the Live window, press Ctrl+Cmd+F, or select Full Screen from the View menu. In Windows, press F11 or select Full Screen from the View menu. You can also use any of these methods to exit Full Screen Mode, and Windows users get a small arrow button at the bottom right of the screen that returns the DAW to windowed mode when clicked.

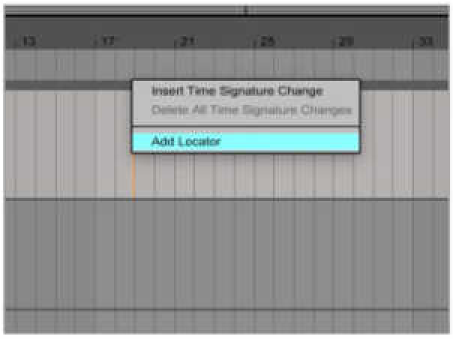
Live also has a secondary window that automatically displays whichever view isn't selected in the main window. It's activated by selecting Second Window from the View menu, or by pressing Ctrl/Cmd+Shift+W.

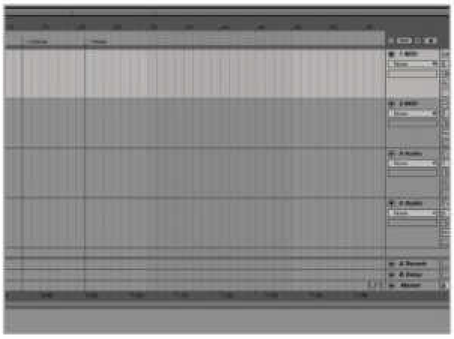
Another way to customise Live's appearance is by adjusting the Zoom Display in the Preference menu's Look/Feel tab. This can be set to up to 200%, which will help those with visual impairments make things out more easily.

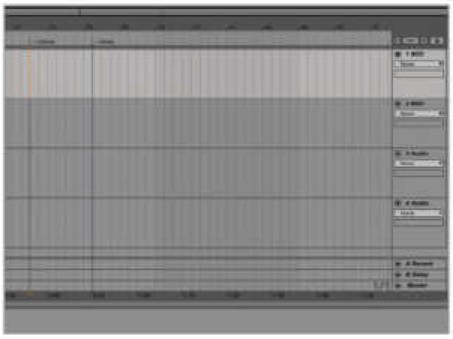


You can make everything tiny or huge with Live's Zoom Display setting

> Step by step 21. Setting locators







1 > Live's Arrangement Locators are markers that enable you to instantly move playback to particular points in the track – choruses or verses, for example. To add a Locator, right-click the Scrub Area and select **Add Locator**.

2 > A Locator will be created and its name field highlighted so that you can enter a new name for it. You can also add a Locator at the current playback position by clicking the **Set** button to the right of the Arrangement. Note that the positioning of locators is controlled by global quantisation, so at the default settings, they'll snap to the nearest bar.

3 > You can move playback between Locators by clicking the arrow buttons on either side of the **Set** button. To select a Locator, click it. When a Locator has been selected, the **Set** button will change into a **Del** (delete) button, which will remove the Locator when clicked. To activate playback from a Locator, simply double-click it, or single-click it if the project is already playing.

Racks and chains

We've already covered Live's Racks briefly in *Audio slicing and creating custom library content* on p18, but being such powerful tools, they're worthy of more comprehensive examination here.

Racks make it possible to create self-contained chains of instruments and effects, which can be saved as presets and very quickly added to new projects. What's more, Racks enable you to create stacks of chains, facilitating the creation of layered instruments and parallel processing effects.

Given all this power, you might imagine that Racks would be tricky to work with, but their sensible design makes it easy to both build and control complex instrument and effect setups.

Chaining and layering effects is a great way to make new sounds, and in Live the possibilities are endless

The Instrument, Drum and Effect Racks all have a bank of eight built-in Macro Controls, each of which can be assigned to multiple parameters. This, combined with the ability to stack chains, makes it possible to create your own composite devices that can do things Live's stock instruments and effects can't. For example, in the following walkthrough we'll show you how to create a formant-style filter using a couple of Auto Filters in an Effects Rack.

Another cool feature of Racks is the ability to restrict chains to particular key and velocity ranges, making it possible to create velocity and note splits – just the thing for live performers and designers of sophisticated stacked instruments. You can also assign a Chain Select range to each chain, and use the dedicated Chain Select Ruler parameter to determine which chains are active at any given time – handy for varying a sound using a MIDI controller knob or automation.

> Step by step

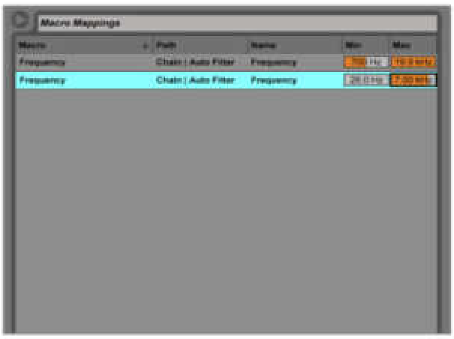
22. Building a formant filter in an Effect Rack

1 > Drag **Hoover.wav** onto an audio track in the Arrangement view, then open **Audio Effects** and drag an **Audio Effect Rack** onto the track. The Rack won't make any difference to the input signal until we add an effect. Drag an **Auto Filter** onto the Rack, and you'll see its interface appear within the Rack.

2 > What the Rack isn't showing you is that it's automatically created a chain for this effect. Click the **Show/Hide Chain List** button on the left-hand side of the Rack, and you'll see a single chain in the Chain List. Drag another **Auto Filter** from the Browser onto the Chain List. This creates another chain.

3 > Because we've got two identical chains, the signal is going to be twice the volume level. To remedy this, turn both **Chain Volume** parameters down to **-6.0dB**. Set both Auto Filters' **Filter Types** to **Bandpass**. Next, we'll set up a Macro Control. Click the first chain to select it, then right-click the Auto Filter's **Filter Cutoff** and select **Map to Macro 1**.

4 > This causes the Macro Controls to appear on the left-hand side of the Rack, with the first knob assigned to (cutoff) **Frequency**. Click the second chain and, again, right-click the Auto Filter's **Filter Cutoff** and select **Map to Frequency**. Now, both **Filter Cutoff** parameters will be assigned to the first **Macro** knob.



5 > When a parameter is assigned to a Macro, its value instantly jumps to that of the Macro. Consequently, both our filters have the same **Filter Cutoff** frequency, but we need them to be at different frequencies for our intended formant filter-style effect. Click the **Map** button at the top of the Rack and, in the Browser, set the first row's **Min** to **700Hz** and the second row's **Max** to **7.00kHz**.

6 > Click the **Map** button again when you're finished to confirm these changes. Now, with their different minimum and maximum ranges, the Auto Filters' **Filter Cutoff** frequencies will never be in the same position, giving us a satisfying formant filter-style sound. You can even make this effect stereo by panning one chain hard left and the other hard right.(Audio: **Stereo filter.wav**)

DJing with Live

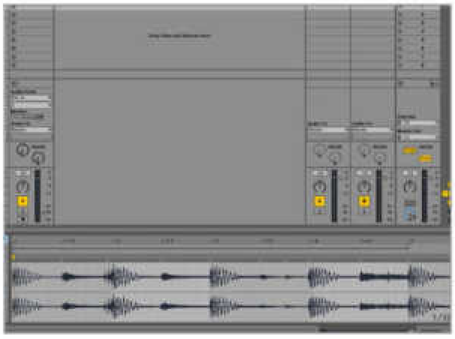
We've already looked at jamming with audio clips in *Using Live's Session view* on p21, so now let's take things a step further by exploring Live's DJing functionality.

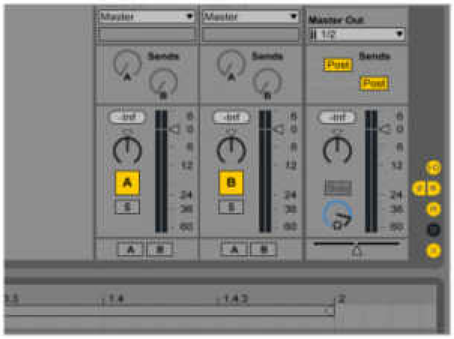
Traditional vinyl-based 'beat mixing' involves manually synchronising two (or more) records so that they play perfectly in time with each other, enabling the DJ to create a seamless mix of continuous music. This technique requires two turntables with pitch controls for adjusting their speeds and a mixer to blend their outputs. Beat mixing with Live works in a similar way, the main difference being that rather than making pitch adjustments on a turntable in real time, audio clips in Live have to be pre-warped before the set starts, so that they stay in time with Live's master clock – for more on this, see *Warping audio clips* on p13.

Although mixing with Live doesn't require you to sync tracks manually in real time, every other process involved is pretty much the same as it is using a traditional turntable/mixer setup, and Live even features a virtual crossfader (the horizontal fader found on any DJ mixer, used to fade from one track into another). Live's crossfader is built into the Session view, which also includes assignment buttons for allocating tracks to either of its two channels. This dedicated DJing functionality makes it easy to get mixing in Live, and the DAW's huge range of audio effects and ability to map parameters to MIDI controllers give it practically unrivalled depth and versatility as a DJing tool. In the walkthrough below, we're going to show you how to 'beat match' two tracks, fade between them, and add a few effects.

You don't need specialised DJing hardware to mix tracks, DJ-style – just a laptop and Ableton Live!

> Step by step
23. DJ mixing in Live

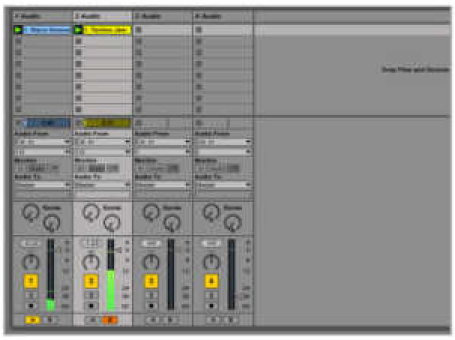


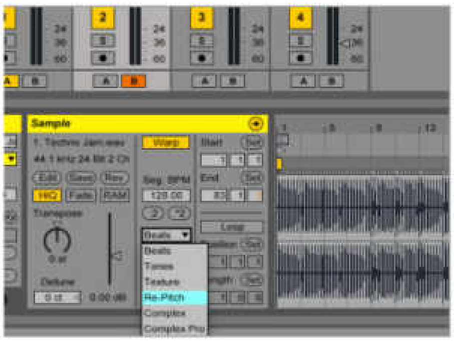


1 > The first thing you need to start DJing is some music, and you'll find two mini-tracks in the **Tutorial Files** folder: **Disco Groove.wav** and **Techno Jam.wav**. Drag them onto Live's Session view while holding **Ctrl/Cmd** – this tells Live to put each clip on its own track. Drop them into the topmost slots of the first two tracks.

2 > It's of the utmost importance that the tracks being played are warped correctly, or they'll drift out of sync with Live's master clock. Thankfully, Live does a great job of analysing these particular tracks: double-click either clip and zoom in on it in the Sample Display, and you'll see that the Start and Warp Markers are positioned perfectly.

3 > Without having to worry about warping anything, we can get straight into the fun stuff. Click **Show/Hide Crossfader Section** – the small round **X** button located at the bottom right-hand corner of the Session view. Live's crossfader will appear at the bottom of the Master channel, and crossfader assignment buttons will appear at the bottom of each track.





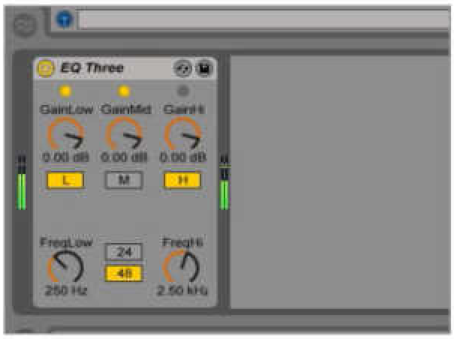
4 > Assign the first track to channel **A**, and the second track to channel **B**. Click the clip launch button on the **Disco Groove** clip. The song will play back, and you can make it louder by moving the crossfader to the left, or quieter by moving the crossfader to the right.

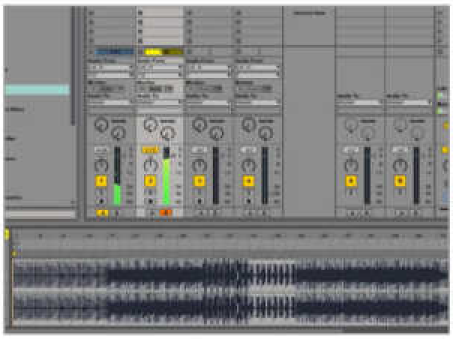
5 > Click **Techno Jam**'s clip launch button. The track will begin playing at the start of the next bar, perfectly in time with **Disco Groove**. Try using the crossfader to smoothly blend one track into another, and to rhythmically 'cut' between the two tracks rapidly.

6 > That's all you need to know to perform simple beat mixing, but we've only just scratched the surface of Live's DJing capabilities. Let's see how we can make this mix sound slicker. For starters, change both audio clips' warp modes to **Re-Pitch**. In this mode the clips are resampled rather than timestretched, making them sound less glitchy but changing their pitches depending on how much they're sped up or slowed down.

> Step by step

24. DJ mixing in Live (continued)

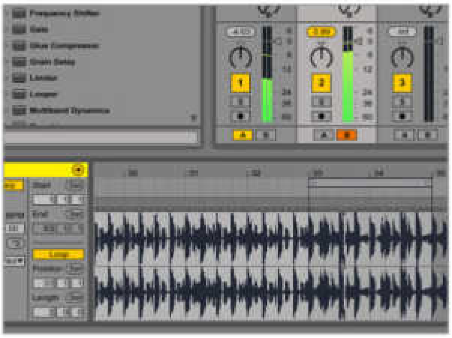


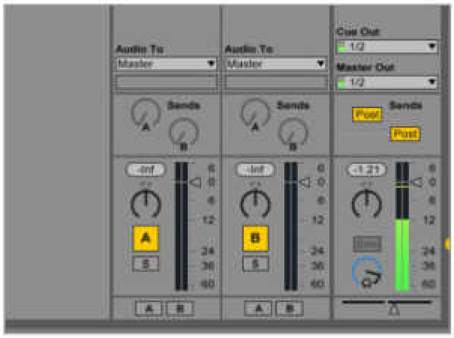


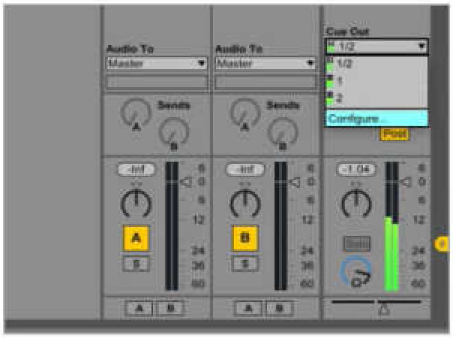
7 > Live features an equaliser that's practically purpose-built for DJing: EQ Three. Drag it from the **Audio Effects** folder onto the second track. It gives us simple level controls for the low, mid and high frequencies. Try turning down the **Mid** knob when **Techno Jam**'s squelchy 'acid' line plays: it makes it almost inaudible in the mix, but keeps the low-end kick very much in evidence.

8 > You can also use the **On/Off** buttons for each frequency band to silence them, and if you're brave enough to mix without a MIDI controller, assigning these buttons to keys on your computer's QWERTY keyboard allows you to 'kill' frequencies much more quickly and easily than using your mouse or trackpad.

9 > Because you're playing a digital file rather than a physical record, you can, of course, skip playback around your tracks as you see fit. Double-click the **Techno Jam** clip to bring up its Clip View. Clicking in the bottom half of the Sample Display tells Live to jump to that part of the track.







10 > Because global quantise is set to **1 Bar**, at the start of the master clock's next bar, Live will automatically jump to the beginning of the bar nearest to your where you click. You can also set up loops in real time by clicking the **Position Set** and **Length Set** buttons, which place loop start and end points, respectively, at the playback position.

11 > Even though Live will keep everything in time as long as your clips are warped correctly, it's still useful to be able to 'cue' the next track up to ensure it sounds good with the one that's currently playing. The **Solo/Cue** button on the Master channel is greyed out by default, because we haven't set up a Cue Output.

12 > The **Cue Out** menu is located on the Master channel, and it can be set to any of your audio interface's stereo or mono outputs. If you can't see the output you want to use in the list, select the **Configure** option to bring up the **Audio** page of the **Preferences** window. Activate the one you want to use in the Output Config area.

13 > With the Cue Output set, the **Solo/Cue** button is no longer greyed out. Click it to activate **Cue** mode, turning the **Solo** buttons into **Cue** buttons. Now, when you click a **Cue** button, you'll be able to hear that track via the Cue Output.

14 > The level of the cue signal is controlled by the **Preview/Cue Volume** knob below the **Solo/Cue** button. This is important because the cue signal is useless if you can't hear it properly, and you may find that it needs a bit of a boost in a live performance context. This knob also controls the level of audio preview in the Browser.

POWER TIP

>Repeat performance

A particularly fun effect to use when DJing with Live is Beat Repeat, which generates stuttered repeats of the input signal. Put it on a track, and set the **Grid** to **1/4**, **Gate** to **1 Bar**, and **Output Mode** to **Ins** (Insert). Click the power button at the top left-hand corner of Beat Repeat's interface to bypass it, then play a clip back on that track and un-bypass it. The effect will repeat the beat at the beginning of the next bar indefinitely until it's bypassed again. Adjusting the Grid while the effect is active can result in some intense glitch-style effects.

Sidechain routing

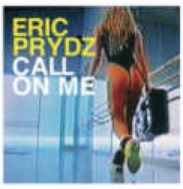
Most of Live's audio processing effects have a single stereo input, but Compressor, Gate, Glue Compressor, Auto Filter, Multiband Dynamics and Vocoder all feature an auxiliary or 'sidechain' input that enables them to accept another audio signal separate to the one being processed. What these effects all have in common (aside from Vocoder, which is a special case that we cover in *Extreme effects with Vocoder* on p142) is that they make use of envelope followers.

An envelope follower detects the amplitude level of a signal and uses it to control the effect processing that signal. For example, a compressor only begins to attenuate the signal once its input has exceeded a certain threshold level; and Live's Auto Filter has an Envelope parameter that determines how much the amplitude of the input signal modulates the filter cutoff frequency.

Sidechain routing involves sending a signal from another track to the envelope follower, so that the amplitude of that track, rather than the main input signal, is the one controlling the effect and defining how it sounds. The most ubiquitous application of this technique is sidechain compression, where the input signal is 'ducked' (attenuated) when a particular sound on another track plays. A famous example of this in action can be heard in the Eric Prydz classic *Call On Me*, where the musical loop ducks in volume when the kick drum plays, creating a pumping effect.

So-called 'trance gate' effects, meanwhile, feed a gate effect with a sidechain input from a hi-hat (or other percussive sound) track to raise and lower the volume of a sustained signal, such as a lead guitar or synth pad, creating a rapidly strobing sound.

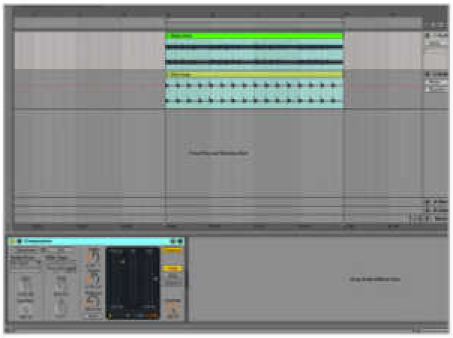
Not all uses of sidechaining are so obvious, however, and it's often applied imperceptibly to allow particular elements of a track to punch through the mix. In this walkthrough, we'll show you how to use Live's built-in sidechaining to help a kick drum and bassline both sound loud and full in the mix.



Eric Prydz' *Call On Me* brought pumping sidechain compression to the charts

> Step by step

25. Ducking a bassline with a kick drum



1 > Drag **Kick loop.wav** and **Bass loop.wav** onto audio tracks in Live's Session View, and Press **Ctrl/Cmd+L** to set up a loop around the clips. Drag a **Compressor** from the **Audio Effects** folder onto the bass track, then click the **Sidechain Toggle** triangle at the top left-hand corner of the interface to bring up the effect's sidechain parameters.

2 > Click the **Sidechain** button to activate the Compressor's sidechain input, then click the **Audio From** drop-down menu to display the list of available audio sources and select the kick loop track. On playback, you'll see the Compressor's **Threshold** meter respond to the input it receives from the kick track.

3 > Drag down on the **Threshold** fader and eventually you'll see the **Gain Reduction** (**GR**) meter start to pump along with the kick drum. This shows us that the Compressor is lowering the volume of the bass whenever the kick plays. Set the **Threshold** to **-20dB**. (Audio: **Sidechain compression.wav**)

4 > The Compressor's **Release** parameter determines how quickly it responds to input level changes. Turning it up gives the kick more room to breathe, but if you don't want to squash the bass too much, a faster setting is the way to go. Setting the **Release** to **20.0ms** gives a good balance between punchy kick and solid bass. (Audio: **Faster Release.wav**)

5 > Solo the bass track. We can tune the sidechain input's response to a specific frequency range with the effect's built-in equaliser. Click the EQ button to activate it, then click the **Sidechain Listen** (headphones) button to hear the EQed input signal. Set the sidechain EQ mode to **Lowpass**. This filters the input signal so that only low frequencies duck the bass.

6 > Set the **Frequency** to **80Hz**, then click the **Sidechain Listen** button to make the bass audible again. Unsolo the bass track to hear how the effect sounds. You can use the **Ratio** knob to control the level of the ducking, and even mix some of the bass into the sidechain signal with the **Dry/Wet** knob in the **Sidechain** panel, making the effect less pronounced. (Audio: **Filtered sidechain.wav**) **cm**

The cm guide to Ableton Push

DOWNLOAD
Get the tutorial videos on your PC/Mac at vault.computermusic.co.uk

Live 9 really comes alive with its own dedicated controller. Let's get hands-on and build a track

> Back in 2013, when we first learnt that Ableton were developing their own hardware controller for Live, with Akai handling the engineering, we were elated and apprehensive at the same time. With Akai's APC40 already serving as an excellent clip-launching and mixing device, clearly Push had to offer something novel and 'different' to find its place in the world – but what would that be, and would it be something we really needed?

The answer to the first question, it turned out, was that Push not only gave us hands-on control of Live's browser, devices and mixer, but was also an instrument, giving us a truly viable alternative playing surface to the conventional MIDI keyboard. While the 'in key' trigger grid concept weren't new, Ableton pulled it off with aplomb, giving trained keyboard players an interesting new instrument to experiment with, guitarists a 'fretboard-like' performance interface that they felt instantly at home with, and untrained musicians a system for MIDI note entry that they found both intuitive and enjoyable.

The answer to the second question was – and is – a little less straightforward. You don't *need* Push, of course, but we certainly wouldn't hesitate to recommend it to any Live user as the best controller currently available for Ableton's excellent DAW. And with its featureset likely to continue expanding in future versions of the software, plus the potential for Max For Live programmers to leverage it in ways yet to be imagined, surely there's much more to come with this already-superb device.

In this tutorial and its accompanying videos, we'll put together a small project using several facets of Push's functionality. From recording MIDI parts and tweaking devices to Racking up a third-party plugin to make it Push-compatible and triggering clips in Session View, no stone will be left unturned. Well, maybe one stone: User mode, which unhooks the unit from Live and sets it to output MIDI notes and CCs. We won't be getting into that here, but check out the Push manual for the full low-down.

> Step by step

1. Step sequencing drums

1 > Let's start by step sequencing some drums. With a new project started and a MIDI track loaded, we need a sound source, so we hit the **Browse** button on Push to hunt for a drum kit. Push can only browse the Live library and locations in the Places browser category for Live devices. Third-party plugins and sounds have to be integrated into it to show up, however, which we'll come back to later.

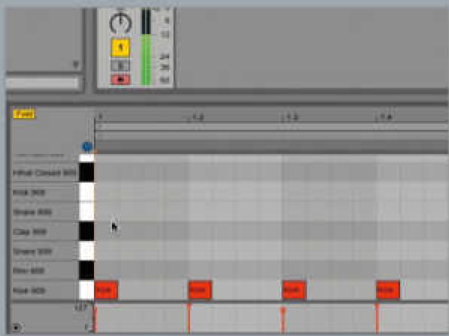
2 > The left hand column of the browser display shows all of Live's virtual instruments – we can scroll through the list using the knob above it, and load one by pressing the green button below. The next column contains the instrument's preset categories (folders), while the two right-most columns reveal the presets in the selected folder. We select **Drum Rack»Core Library»Kit-Core 909**.

3 > In drums mode, the bottom left 4x4 block corresponds to the drum sounds in the Rack – if your kit has more than 16 sounds loaded, you can scroll through them using the touchstrip. The top four rows comprise the step sequencer, with the step resolution determined by the Grid buttons on the right. We start playback and enter kick drum hits on every beat of the loop.

POWER TIP

>Trigger unhappy

So, you're step sequencing or recording a drum pattern live on stage and you want to switch the active drum from the kick to the snare. Surely striking the pad to do so will trigger its sound, right, causing much embarrassment? That's where Push's Select button comes into play: holding it down while hitting a drum pad switches to that sound for note entry or live triggering without actually sounding it.



4 > The loop length is set using the bottom-right 4x4 block of pads. Starting at its top left, each lit pad represents one bar. At the moment we have a two-bar loop: the green button shows which bar is currently playing, while all other active bars (or bar, in this case) are lit up. If we were to hold down the top left pad and hit the bottom right one, we'd have a 16-bar loop.

5 > At the moment, all our kick drum hits are at the same velocity. Pressing the **Accent** button sets the velocity of newly entered notes to maximum. Here we're upping every other kick drum to maximum velocity before returning them to the regular level.

6 > Next, we place a snare and handclap on the backbeat. At the default velocity level, the snare is a bit quiet, though, so we press the **Accent** button on the right-hand side of the unit and reenter the hits to set them to maximum velocity.

7 > Holding down a pad in the step sequencer gives access to four editing functions for that note via the rotary encoders. The first is **Nudge**, which enables us to shift the selected hit left and right. The second and third are **Length** and **Fine**, which don't work for our one-shot clap sound, but would shorten or extend a sustained sound. Finally, the third encoder adjusts the **Velocity**.

8 > Now for some hi-hats. We tap in closed hats on the offbeats and an open hat on the last offbeat of the second bar. Individual drums can be soloed and muted: holding down the **Solo** button and hitting the kick drum pad solos the kick, for example, while holding down the **Mute** button and hitting the same mutes it.

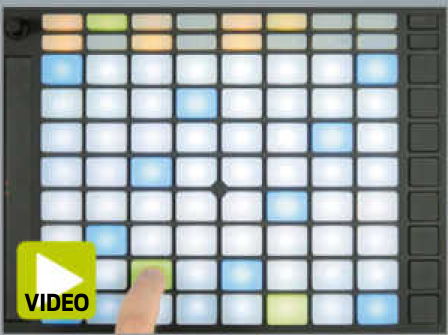
> Step by step 2. Real-time drum recording

1 > Push also facilitates live recording of Drum Rack parts, MPC style. We press the **Add Track** button to add a new MIDI track. The Browser opens in the Push display. We're going to record some percussion, so we select **Drum Rack»Core Library»Kit_Percussion**. Pressing the green button loads our chosen Drum Rack.

2 > You can freely record a part of any length you like, but since our drum loop is two bars long, we'd like our percussion to match it automatically. Holding down the **Fixed Length** button presents us with a range of musically logical clip durations to choose from. We opt for **2 Bars**, then hit **Record** and perform our part live on the pads.

3 > Push's Repeat function enables rolls of varying speeds to be generated by holding down the pads: we turn **Repeat** on and hold down the conga pad to generate an eighth-note roll. The Grid buttons to the pads' right set the Repeat rate, and the pressure applied to the pad controls the volume of the repeats. We activate record again and capture our roll.

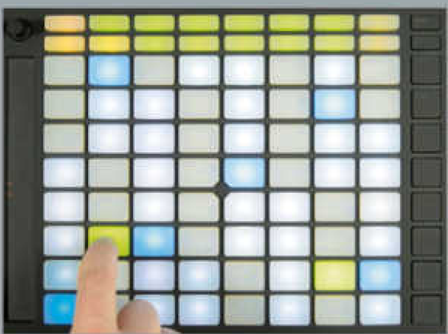
> Step by step 3. Recording a bassline

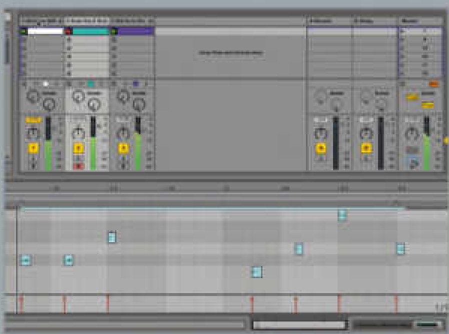


1 > We press **Add Track** and load Analog's **Dual Osc2 Rubber Bass** preset. The way Push handles pitched instruments is one of its defining features. The default **In Key** mode only activates the notes of the currently selected scale, with blue pads representing root notes and notes stepping upwards within any three-pad-wide column. The default is C Major.

2 > Each note is repeated several times across the pad grid, and pressing one will make all equivalent notes light up in green. Pressing the **Scales** button lets us set the grid to any one of a comprehensive range of keys – major, minor, numerous modes and lots more 'exotic' options. We go for **F minor**.

3 > In Scale mode, holding the **Shift** button enables reconfiguration of the pad layout. By default, each pad plays an interval of a 4th above the one below. This can be switched to a 3rd, and the whole arrangement can be rotated by 90° so that up and down become right and left. You can also set the pads to a continuous sequence of pitches over eight octaves.





4 > In Key is Push's default mode, filtering the pads down to just the notes in the scale of the current key. There's also your regular **Chromatic** mode, though, if required, in which all the notes of the current key are lit but all the other notes are also present, unlit. We stick with In Key mode.

5 > We start playback, hit the **Record** button and play in a bassline. Bum notes are easily removed with a press of the **Undo** button, and hitting the **Quantize** button quantises our newly recorded notes to the current grid as determined by the Grid buttons. A press of the **Note** button switches to the step sequencer view for detailed note editing via Push.

POWER TIP

>Lost your keys?

What you should remember about Push is that its state isn't saved either in the unit itself or as part of your Live projects. This means that everything resets to default when it's powered down, including, most pertinently, the Scale setting. So, it's definitely a good idea to make a note of which key you're in for each project, otherwise you're going to have to work it out for yourself every time you load a new one or power cycle Push.

> Step by step

4. Automation, third-party plugins, effects and mixing

1 > Push's rotary encoders can edit and automate instrument/effect parameters such as the filter Cutoff and Resonance of our bass synth. The top row of buttons (State Select) switches through banks of parameters: oscillators, filters and envelopes, for example. To record parameter edits as clip automation, press the **Automation** button, enter **Record Mode** and twiddle those knobs!

2 > Third-party plugins don't appear in the Push browser and can't be controlled via the rotary encoders without some prior setup. We hold down the **Add Track** button to bring up a choice of Audio, MIDI or Return track, and select **MIDI**. A MIDI track is created in Live. We're going to use Native Instruments' **Massive** synth, so we search for it in the Live browser.

3 > We could assign controller mappings in Massive for this project using the standard Live plugin Configure button, but for a long-term solution, we can wrap it up in an Instrument Rack and save it into the Library. To do this, we right-click its **title bar** and select **Group**. Our Instrument Rack has eight Macro knobs. Massive has eight Macro knobs. And Push has eight rotary encoders! How very convenient.

4 > Clicking the main Map button in the Rack enables us to assign its eight knobs to selected parameters in Massive – specifically, the Macros. We click each Massive **Macro**, then the **Map** button underneath the target Rack Macro to make the assignments. With Live's Macros mapped to Massive's Macros, we're well set up to control the synth with hands-on parameters tailored to each preset.

5 > The last step is to save our Rack into the Live Library. Simply click the **Save** button in the Rack and give your saved Rack a name. Now you can load your Macro-mapped Massive Rack from the Library whenever you need it, either onscreen or, more importantly, via the Push browser.

6 > Now to record a pad sound. We're in Push's In Key mode, so all we have to do to trigger a series of related chords is make a shape with our fingers and move it around the grid. To record a series of clips in succession, the **New** button copies all the clips in the current Scene into the corresponding slots in the next Scene down, except for the selected one (our Massive pad) which is record-armed.

7 > Live's Effects can be loaded/controlled via Push, too. With our Massive track selected, we press the **Add Effect** button, choosing Auto Filter's **Elastic Band** preset. Pressing the **Device** button switches to Device mode, mapping the Auto Filter's controls to Push's rotaries. The **State Control** buttons bypass devices, while the **Selection Control** buttons select them for editing using the rotaries.

8 > Push offers basic mix control via its rotary encoders, too. Press the **Volume** button to assign them to per-track level control. Pressing the **Pan & Send** button repeatedly switches the encoders through Pan and as many Sends as you have set up. The **Track** button switches to Track View, in which the rotaries are assigned to the Volume, Pan and Sends for the currently selected track.

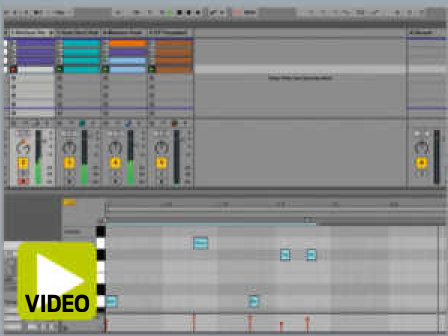
9 > The **State Control** buttons handle mute and solo: mute with the **Mute** button engaged, and solo with the **Solo** button engaged. Finally, the **Clip** view gives you control of various Clip parameters such as start point, length and Loop on/off.

> Step by step

5. Recording a Rhodes riff

1 We elect to add a melodic part to our track in the shape of a Rhodes emulation from Live's **Electric** instrument. Loading the **E.P. Faceplant** preset, we feel that our Rhodes is a bit quiet, so we start by turning the volume up. Staying in F minor, we jam out a short riff.

2 We hit the **New** button to prepare a new Scene and **Record arm** a second Rhodes clip. This time we're just going to jam around until we get something usable, so we turn off **Fixed Length** and get busy improvising.



3 Let's wrap up our little demo track by recording a few alternative parts. First we'll record a couple of new Massive clips. We press the **New** button, enter record mode and make a chord shape, then do the same again to make a third clip. Then we record a new percussion clip and quantise it.

4 Switching into Session mode changes the pads from note and drum triggers to clip launchers, coloured to match the clips in our project. The Navigation buttons move the 8x8 focus window around the project.

5 Holding down the **Shift** button zooms out to show an overview of the Session, with each pad switching the focus to an 8x8 section, enabling fast navigation of even the biggest project. Check out the **More Session mode functions** bonus video on the right to see this in action.

6 Let's trigger some clips! In Session mode, hitting a pad triggers the corresponding clip, obviously, while the Grid buttons become Scene launch buttons – it's as simple as that. Pushing the **Stop** button assigns the State Control buttons to the Clip Stop function, enabling the clips playing on each track to be stopped. cm

Pushing the boundaries

Push is a pretty intuitive beast with a very finite range of functionality that doesn't take long to get comfortable with – the workflow soon becomes second nature and you quickly find yourself flying around the thing. Here are some handy tips and tricks...

- **Shift+Play acts like the Live Stop button, stopping playback on first press, returning the project to the start on second press, and killing all audio and MIDI on a third press.**

- **Many of Push's buttons have 'hold' functionality as well as their regular on/off switching. For example, in Session mode, press and hold Note to temporarily switch to Note View for MIDI editing, then release it to go back to Session mode. The reverse also works.**

- **Press Shift+Add Effect to bring up Live's MIDI effects in the browser, ready for navigation and loading just like their audio equivalents.**

- **The first of the two footswitch inputs on the back of Push supports a sustain pedal, while the second can be fed a pedal input for foot-operated activation and deactivation of record mode. Stomp once to toggle into record, and again to drop back out into playback mode. A fast double-tap of said pedal, meanwhile, does the same thing as pressing Push's New button, killing playback of the current clip and preparing to record a new one.**

BONUS VIDEO

See how Push can be used to control Live's Session View in this bonus video – download it along with the rest of the videos at **vault.computermusic.co.uk**

DRUM 'N' BASS

TRACK-BUILDER

You don't need to spend big bucks to get that big DnB sound – follow our guide and build every element of a pro-quality track in Live using free plugins and samples!

> Since its inception in the early 90s, drum 'n' bass has always been about using limited equipment in exciting new ways. Although software DAWs as we think of them today didn't exist back then, many producers – including leading lights such as Omni Trio, Aphrodite and Bizzy B – started out with simple tracker applications that enabled them to create entire pieces of music using just a home computer. Despite being created in inexpensive software like Teijo Kinnunen's OctaMED (which was even given away for free with Amiga magazines back in the day), tracks like Urban Shakedown's *Some Justice* and Omni Trio's *Mystic Stepper* pushed the boundaries of dance music, and became hugely popular with DJs and audiences alike.

20 years later, DnB has become very much part of the mainstream, but its innovative sound is still driven forwards by bedroom producers blessed more with passion than expensive kit. Thankfully, though, music production software has moved on since the days of OctaMED, and there are now countless exciting free plugins available to musicians on a budget – not to mention, of course, the amazing Ableton Live, which has become a firm fixture on the drum 'n' bass production landscape. Make no mistake about it, with a little ingenuity, you really can craft pro quality tracks without breaking the bank.

In this start-to-finish tutorial, we'll show you how it's possible to make a full-on liquid DnB roller using only Live, a selection of freeware plugins that you can download from the net, a handful of exclusive **cm** Plugins included with this very magazine for free, and a range of choice sounds from classic *Computer Music* sample packs, including the Drum & Bass All-Stars pack, which can be downloaded at no cost from **bit.ly/dnballstars**. Feel free to use these royalty-free sounds in your own productions beyond this tutorial, too, of course.

The project file for our finished number is also included in the **Tutorial Files** folder, ready for you to load into Live and peruse at your leisure. This contains all the raw, unbounced tracks involved, so you can see and hear for yourself how each and every sound is processed, and how the track is mastered to give it that professional finish and sheen.

Without further ado, then, let's step to it and make some noise...

> Step by step 1. Building a DnB beat in Live with free samples and plugins

1 > We pick **Kicks (20).wav** and **Snares (16).wav** (from the DnB Allstars pack, included in the **Tutorial Files** folder). These are tuned appropriately and don't have long tails, so we can place them in a typical two-step pattern without adding fades to control their tails. Voxengo MSED is used to mute their sides signals, making them mono. (**1. Kick and snare.wav**)

2 > The snare has a little unwanted low-end energy, so we use Philta CM to high-pass it at **145Hz**. CM-EQUA 87 is then brought in to boost the **200Hz** region a little, enhancing the snare's punch, and cut the harsh region around **2kHz**, giving us a smoother feel and making room for other high frequencies in the mix.

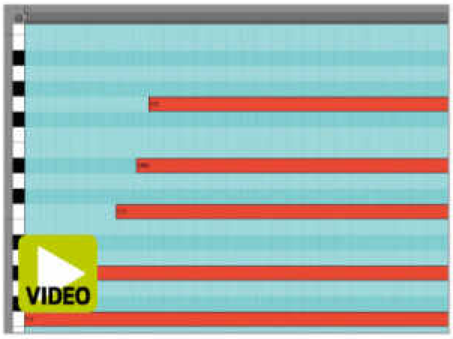
3 > We want a more natural-sounding snare for our liquid roller, so we layer it with one cut from **DnB_Break_29.wav**. CM-COMP 87 is applied to this track, and the long **Attack** time allows the transients to punch through. The two snares sound good together, but currently they're very dry. (**3. Dry snares.wav**)

4 > We layer the snares with **002_D&B_Clap.wav**, which is run through ReverberateCM's **Long Plate** preset. To make the reverb tail cut off quicker, the Transector CM transient shaper is inserted, with **Release Gain** set to **-6dB**. Finally, Philta CM high-passes the signal at **286Hz** to make room for the main snare to punch through at around 200Hz.

5 > Next, **Closed Hats (3).wav** on eighth notes. This sample is stereo but louder on one side, so we make it mono with MSED, then use TAL-Chorus-LX (**kunz.corrupt.ch**) to dial in the stereo width as needed. We cut out everything below **1722Hz** with IIEQPro's HPF12 filter, and add **Open Hats (4).wav** (faded out at the end) at the end of every four bars.

6 > The **Jungle Break 140.wav** breakbeat features a piercing shaker that gets two treble cuts in IIEQPro, which we also use to high-pass the low-end at **68Hz**. The second break, **DNB_BREAK_29.wav**, is tamed in the 2-5kHz region with CM-EQUA 87, then cut up across four tracks, so that different high-pass settings can be used on the kicks, snare and hi-hat sections. All the drums are grouped on a bus channel.

> Step by step 2. Jazzy DnB electric piano with free plugins

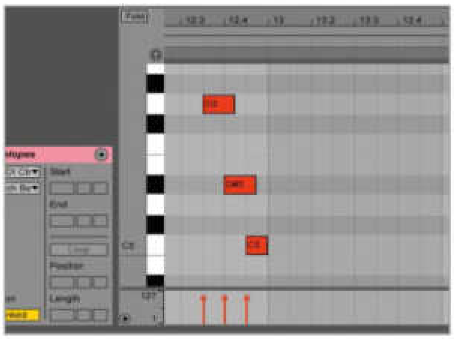


1 > To create a suitably jazzy vibe, we sequence mda ePiano (**mda.smartelectronix.com** or **sourceforge.net/projects/mda-vst** for the 64-bit Mac version) to play a four-bar F minor ninth chord (F2, G♯2, C3, D♯3, G3). For a natural feel, we turn snap-to-grid off and change the start points of the notes so they don't play at the same time. (**1. Dry ePiano.wav**)

2 > To give our chord a silky smooth vibe, we apply MeldaProduction's MPhaser (part of the MFreeEffectsBundle from **www.meldaproduction.com**) subtly, with a **30% Dry/Wet** level. This is followed up with TAL-Chorus-LX to give it extra warmth and stereo width. ReverberateCM's **Hallway 1** preset is brought into play to provide a lush reverb.

3 > Finally, our electric piano is run through CM-EQUA 87, which cuts out the sub-300Hz frequencies and dips the 340Hz region. A bus channel is created for this and subsequent keyboard parts, which will be used to sidechain-compress them later. (**3. ePiano EQ.wav**)

> Step by step 3. Slick DnB leads with Dune CM and free effects

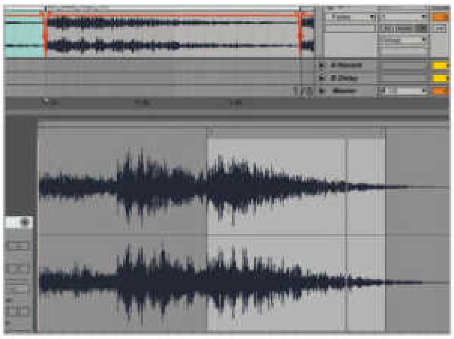


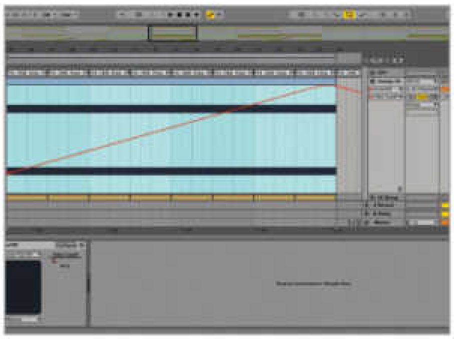
1 > For our synth lead, we use Synapse Audio's flexible Dune CM synth. The patch is based on a single oscillator, with the harmonically rich waveform **11** selected via the **SEL** oscillator mode. The **Fat** knob is turned up a little for a touch of unison detune, then **Mono** mode enabled and the **Glide** turned up to **41%** for a lazy portamento sound. (**1. Dry DuneCM.wav**)

2 > TAL-Chorus-LX boosts the stereo width of the synth, which is then run through our own Wolfram CM, with high **Feedback** and **Wet/Dry** levels creating an epic synth wash every time the riff plays. The chorus and delay effects make the sound very wide, so we employ Voxengo MSED (**www.voxengo.com**) to turn down the **Side** signal slightly.

3 > The synth needs to be bright to cut through the mix, but it's a little *too* bright, so IIEQPro is brought in to high-shelf **-1.3dB** at **1360Hz**. Here, you can see the overlapping MIDI notes we've used to introduce the glide, giving the part its slinky sound. The part plays at the end of every four-bar section, with a variation every 16 bars. (**3. DuneCM EQ.wav**)

> Step by step 4. Lush, layered DnB pads and strings from free plugins

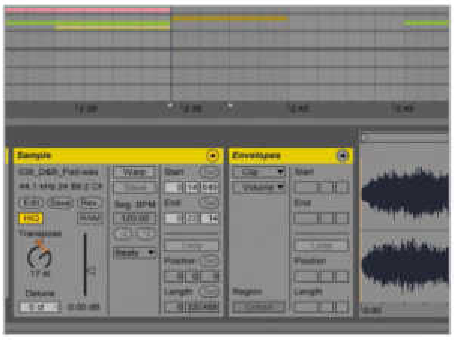




1 > Our first pad is **002_D&B_Pad.wav**, which plays every two bars. We play this from the start of the sample, and its slow attack time gives the sound a sidechain-style ducking effect. It's tuned up **5** semitones to sit with the electric piano part, then high- and low-pass filtered with Philta CM to occupy a narrow band in the mix. (**1. Pad 1.wav**)

2 > The second pad, **034_D&B_Keys_Riff.wav**, is a Roland JV series-style "auto-wah" electric piano chord. The complete loop is a phrase consisting of three chords, and we've chopped just the last one out as it's in the right key for our track. Short fades are applied to ensure that the start and end of the sample don't click.

3 > The long, sweeping pad is a Dune CM preset, **Hollywood Pad RH**, playing a perfect fifth. There's nothing particularly complex about this sound – it's just a sawtooth pad with slow **Attack** and **Release**, and a little unison detune. The filter **Cutoff** is slowly automated up over 16 bars, developing the sound and increasing in intensity as it does.



4 > The high string sound is created from scratch using a unison-detuned sawtooth oscillator in another Dune CM. To turn it into a natural-sounding string, it's run through Acon Digital CM Verb, the dry signal of which is turned off entirely, leaving just the 'room' sound. Philta CM cuts out any unwanted frequencies below the **1355Hz** mark. (**4. High string.wav**)

5 > The final pad, **038_D&B_Pad.wav**, is pitched up 17 semitones to complement the high string. This plays every two bars, and during one of the breakdowns, Philta CM is used to take out its higher frequencies, shifting gears downwards in preparation for the more minimal second drop. All the pads are routed to a dedicated bus. (**5. Pad 3.wav**)

POWER TIP

>Padding it out

Getting a smooth, full pad sound is one of the more challenging aspects of making a musical DnB roller, and sound selection plays a big part in getting that right. While virtual analogue synths are great for warm, retro-style pads, for that 90s-style digital sheen, a sample-based synth like Omnisphere or Korg's M1 is a good bet; on the freeware front, try Greenoak's Crystal (**www.greenoak.com**), which isn't sample-based, but has the right kind of character. Samples themselves can be a great source of pad sounds, but be prepared to spend a while auditioning them and tweaking tuning and EQ settings to get the desired results.

> Step by step 5. Big, bad DnB sub bass with free plugins

1 > Let's synthesise a big, sexy bass with a freeware favourite: Togu Audio Line's TAL-NoiseMaker (**kunz.corrupt.ch**). Starting with the default patch, locate the Master panel and turn the **Sub** knob all the way down so we can just hear Osc 1. Set Osc 1's waveshape to **Pulse**. Turn the **PW** (Pulse Width) down to **-0.13**. (**1. Pulse wave.wav**)

2 > This gives us a rough tone, ripe for filtering. Open the Synth 2 panel, and, in the Filter section, set **Key** to **0.50**, so that the filter cutoff follows the note played. Set **Cutoff** to **0.32** and **Reso** to **0.20**. Now we've got a simple sub, but it's not particularly exciting, so let's use an LFO to modulate the filter cutoff.

3 > In the LFO 2 panel, set the destination to **Filter**, activate the **Sync** and **Key Trig** buttons, then set the **Rate** to **1/4T** and **Amount** to about **0.2**. This gives us a big, full-frequency wobble bass – but it's too much for our mix: we're really just interested in the low end.

4 > Add Philta CM on the TAL-NoiseMaker channel, and set the **Lowpass** filter to **127Hz**. Now we have a big bass sound, where the first few harmonics are given room to breathe but everything above them is attenuated, giving the snare, breaks and music elements free reign to fill the mix. (**4. Smooth sub.wav**)

Adding sparkle

With the drums, bass, pads, keys and vocal sorted, we've got all the major elements of our track together – but it's the little extra touches that will help it stand out from the crowd.

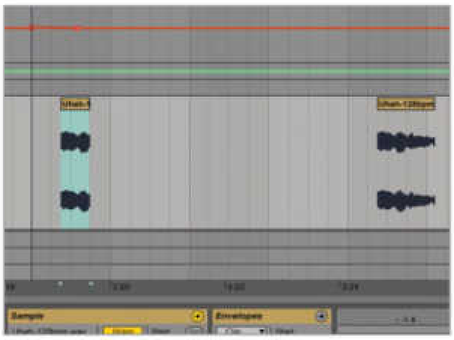
A small but important element that we've added is the harp gliss lead from Alchemy Player CM (it's the Glissering Harps preset), which plays a simple, octave-skipping rhythmic part for most of the track. This helps emphasise the rhythm and creates a harmonic contrast with the ascending bassline.

Cymbal crashes (003_D&B_Crash.wav) mark 16-bar sections, and the tom fills (Tom 1.wav) at the end of every 16 bars (accompanied by a subtle variation of the main beat) provide a switch-up that helps the next 16 bar-section sound fresh when it comes in.

Last but by no means least, pitched and white noise risers created with TAL-U-No-62 lift the energy at crucial moments (usually before breakdowns and drops) and 'boom' FX (DNB_FX_1.wav) mark the start of each breakdown and punctuate the end of the track.

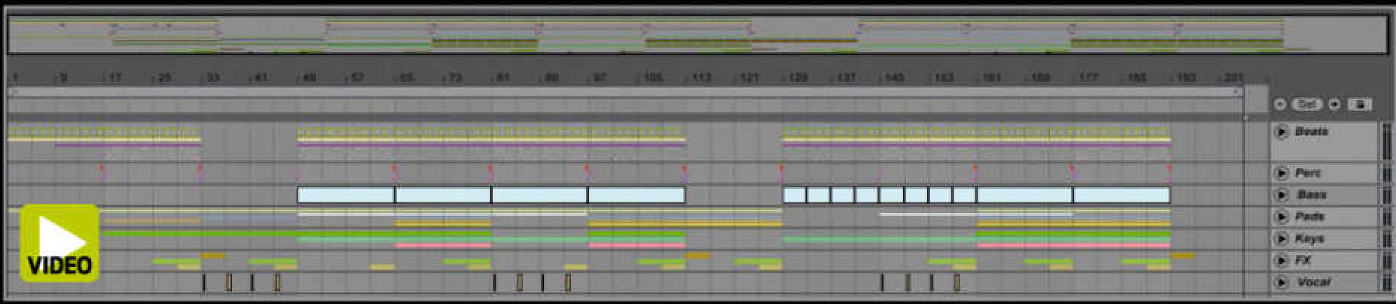
> Step by step 6. Working vocals into a DnB track using free plugins

1 > When working with vocal samples, you can chop and change them a bit from their original form to get something fresh sounding. Here, we've taken **Uhah-128bpm.wav** and stretched a snippet of it using our DAW's timestretching system so that it fits the rhythm of the track. (**1. Stretched vocal.wav**)



2 > Our snippet plays every four bars, but it gets a bit repetitive, so we cut the first and third instances shorter to create a little variation: an "uh-woh" version, and an "uh-woh-ooh" version. Small touches like this can help a track flow and keep the listener interested.

3 > The middle of the mix is dominated by drums and bass, so we move the vocal into the sides of the mix with TAL-Chorus-LX, and use Wolfram to add a quarter-note delay. Finally, MSED is summoned once again to turn down the side signal and make the channel slightly less wide. (**3. Wet vocal.wav**)



7. Arranging a slick DnB tune made with free plugins

At 192 bars, our project is relatively short for a non-'radio' track, but it's pretty DJ-friendly and it sticks to the conventions of dancefloor DnB.

Things kick off with a beat-driven intro, where the single-hit drums play with just a pad for accompaniment – this will make it very easy for DJs to cue up the track and mix it in without things sounding messy.

The extra breakbeats, string and harp are introduced to ramp up the tension, and at bar 33 we arrive at the first breakdown. During this section, the pads and harp filter in, and at bar 49 the beats, bass and electric piano drop.

We leave out certain elements to give the mix somewhere to go, and the next 16 bars see the introduction of the high pads and synth leads – this is the crescendo of the track. Subsequent 16-bar sections see elements added and removed to give the track a little light and shade, building to a second breakdown, where we downshift to a more subdued, pad-less drop, in which a simpler version of the bassline gives the listener a break from the intensity and the DJ an opportunity to mix in another track.

After this, the track shifts back to high-energy mode, giving the listener another chance to hear all the elements before building to the explosive ending.

The track could be made more DJ-friendly by winding down to a beats-only outro before it ends, which would allow the DJ to bring in the next tune more gradually.

Watch the final video or listen to **Master.wav** to hear our finished drum 'n' bass track!

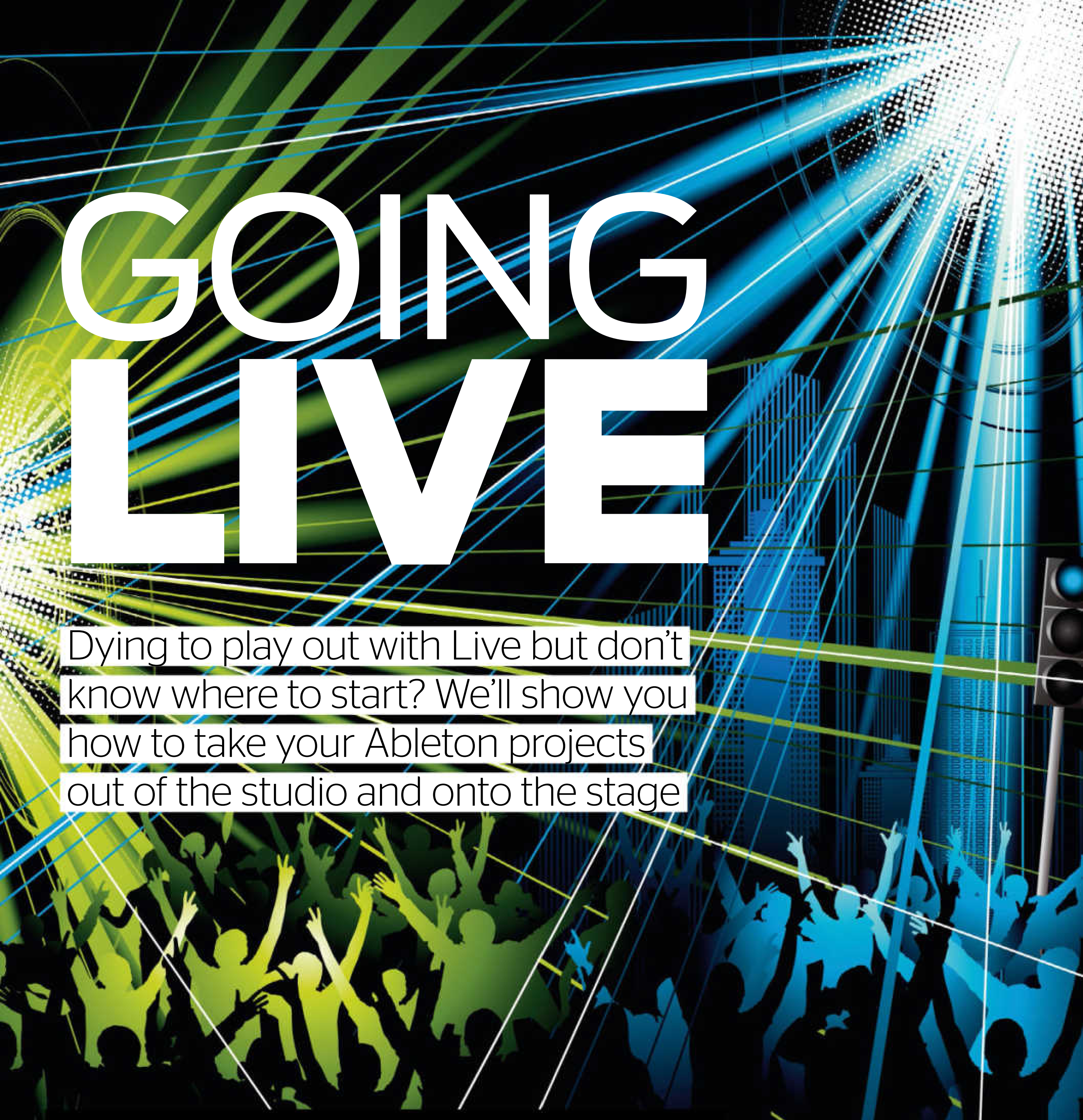
8. Mixing and mastering a DnB track with free plugins

Like most electronic music made today, most mixing tasks will occur throughout the production process. For example, once the drum parts were created and routed to a bus, we used eaReckon's CM-COMP87 to 'glue' the sounds together. Maintaining transients was a concern, so we used a long, 225ms attack time, and a Wet/Dry mix of roughly 50% to ensure we didn't take too much punch out of the mix.

Another way to get the drums punching through is to sidechain some track elements off the kick and snare channels. Although there are some free sidechain compressors out there, routing them is inconvenient, and Live has a fantastic one built in, so this is one situation in which using freeware doesn't really make sense.

At this point, it became clear that the high end of the drums was too harsh, and that the balance between the kick and bass wasn't perfect. These are probably the two hardest elements of any drum 'n' bass track to get right. We tamed the highs by inserting a gentle bell EQ into the drum bus and softening the 5-10kHz region, then turned down the kick slightly to stop it from popping through the mix so much.

When it comes to mastering, getting a loud, fizzy, punchy – but not harsh or distorted – final sound is the Holy Grail of drum 'n' bass production. First, **cm** Plugins' STA Enhancer was applied to excite the mix, then a couple of instances of DDMF IIEQ Pro CM were brought in to get a more even tonal balance. Live's Group function and Voxengo MSED were deployed to split the signal into mid and side channels, each of which is boosted using Barricade CM. After that, the whole signal was run through another Barricade CM to get it as loud as possible without clipping – job done! **cm**



GOING LIVE

Dying to play out with Live but don't know where to start? We'll show you how to take your Ableton projects out of the studio and onto the stage

> **A major contributing factor to Ableton Live's success when it first launched in 2001 was its innovative Session View, a gloriously non-linear loop playback and manipulation interface via which rough ideas could be knocked about as the basis for full arrangements, and – more importantly, it turned out – full tracks and samples could be mixed by DJs into fluid, wildly creative live performances that went far beyond anything that had ever been possible using traditional turntables or CD decks.**

But the Session View isn't just for mixing complete stereo tracks into each other – it can also be used to take total control of finished Live projects, enabling on-the-fly rearrangement and remixing, whether for your own edification or that of the baying crowd. The software keeps everything in sync at all times, so nothing can go fundamentally wrong (in terms of timing, at least), and you can even record your real-time performance as a linear arrangement for further editing and development.

In this tutorial, we're going to take a moderately dense Live arrangement, transfer it to the Session View and jam around with it to turn it into something new. With the project involving a fair few virtual instruments and effects, we'll start by bouncing it down as a set of audio tracks and stems in order to ensure stability and responsiveness – we don't want a rogue plugin bringing Live down mid-gig, after all. Then, we'll improvise an extended remix of the track by triggering Scenes and clips, track muting and much more. We'll even have a go at recording some entirely new audio and MIDI-triggered elements and working them into the track, all without any break in playback.

DOWNLOAD

Watch tutorial videos and hear audio examples at **vault.computermusic.co.uk**

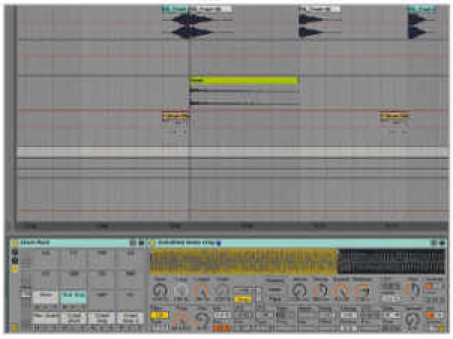
> Step by step

1. Preparing the session

1 > Our track is a stomping, bassy number with a breakdown and drop in the middle, and a general sense of build-up throughout. The drums and bass are built on a range of sources, and we've already marked out our structure using Live's Locators. So that we can play the track out live, we want to get the whole lot into the Session View, mostly as audio clips but with a few parts left 'live' for jamming.

2 > The first stage is to export each channel as a full-track-length audio file – a process that Live makes light work of. We turn off the master channel effects (we'll run those live), set the loop range to encompass the whole track, hit **Cmd/Ctrl-R** to bring up the Export Audio dialog, select **Individual Tracks** in the Rendered Track dialog at **24-bit**, click **Export** and choose a folder for our renders.

3 > We've rendered our drums as full tracks, but it's always handy to have some deconstructed drum loops around, too, so we also export four bars of the full drums mix in solo kick, kick and hats, kick and snare, and hats and snare versions. Similarly, grabbing eight bars of each of our three main bass layers separately will expand our jamming options.



4 > It's also a good idea to give yourself instant access to any crashes, whooshes and other spot FX by loading them into a sampler or Drum Rack on a new MIDI track, even if you keep the full rendered tracks in the project too. We do exactly that now with our crash and siren samples. We'll come back to the MIDI-triggered impact and drum fill shortly.

5 > We save a new version of the project and delete all tracks except the gated synth lead, LFO-modulated 'lead' bass, pad and Spot FX Drum Rack. Then, we simply import our rendered tracks and Group stems (but not the individual tracks that are already in Groups, since we don't want them in our project twice). We name each track and set suitable Warp Modes.

6 > Next, we delete all sections in which there's no sound, as we don't want to create Scene clips for these parts, but we leave the entire 'block' in place in all sections that contain any signal at all. Now we have an audio (plus three synths and a Drum Rack) version of our track. We also chop out the MIDI-triggered spot FX mentioned earlier and put them in the Spot FX Drum Rack.

7 > All that remains is to turn our song sections into Scenes: set the loop braces around each section in turn and select **Consolidate Time to New Scene** from the right-click menu. Voila – the Session View now reflects the Arrangement such that if we play all the way through each Scene from one to the next, it'll sound identical – once we've turned the master effects back on, of course.

8 > Every clip is set to Loop by default, which is fine for all but the final Outro Scene, which ends on a downbeat and thus won't loop properly – so we disable Loop for all of its clips. Then we put our deconstructed drum and bass loops on the Drums and Main Bass tracks, below the full renders, and, finally, colour the clips on each track to indicate variations.

POWER TIP

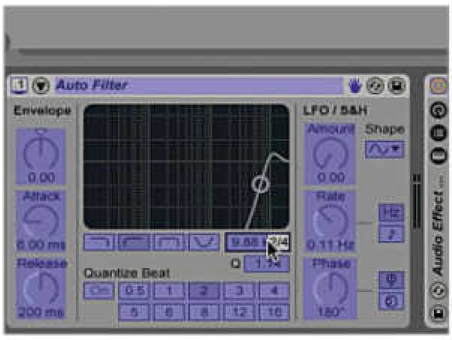
>Staying with sends

In rendering all individual tracks, we also exported the two effects returns (reverb and tape delay), but we've not actually loaded them back in since we'd prefer to keep the send effects running live, giving us the option of sending other tracks to them should we feel the urge. To achieve this, we simply use the same send levels on the rendered versions as on the original tracks – in this particular case, there are only two of them: the rhythm guitar (delay) and the 'Bells' percussion track (reverb).

> Step by step 2. Setting up for MIDI control and recording

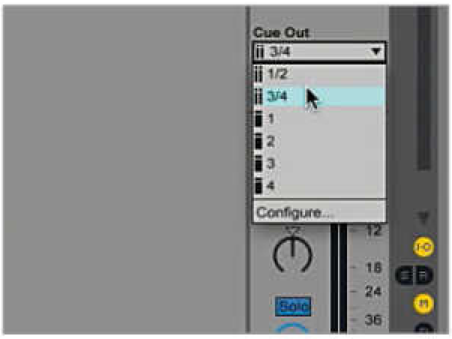
1 > As well as jamming around with our track by clip triggering and mixing (for which we're using Ableton Push and our mouse), we also want to trigger our Spot FX Drum Rack – as well as any other instruments that we might choose to load in the heat of the moment – and control a filter effect on the master output (for big sweeps) using a MIDI controller keyboard.

2 > With a keyboard hooked up, we set the Spot FX channel's **Monitor** field to **In**, leaving it permanently open to keyboard input without the need to select the track first. We turn **Monitor Off** on all other MIDI tracks so that we don't inadvertently trigger them, too, when they're selected.



3 > We want to be able to bypass our master filter at the press of a button and control its cutoff using a knob. Enter Live's **MIDI mapping** mode and make assignments by clicking each parameter and moving the required control on the MIDI keyboard. We could set up other synth and effects parameters in the same way, of course, but we'll leave it at that for now.

4 > Let's incorporate an external hardware instrument (Korg Wavedrum) into our jam, recording it on the fly. We'd like to monitor via headphones on a separate cue output (for editing and creatively manipulating the recordings without the audience hearing us do it). Any audio interface with at least two stereo outputs will work – ours is a Focusrite Forte.



5 > We set the Master channel **Cue Out** to the stereo headphones' output (**3/4**), as set up in Forte's control app. Now we have the ability to take tracks out of the main mix (or not) and send them to the headphones for 'silent' monitoring (having first switched from Solo to Cue mode using the Solo/Cue button on the Master channel). We'll come back to this later...

POWER TIP

>Monitoring: the situation

How your particular audio interface handles setting up cue monitoring is something you'll need to turn to its manual to figure out, but there'll be some sort of monitor mixer in its control panel or setup app that enables you to route the outputs from your DAW to physical outputs on the interface itself. Tell it to send whichever Cue Out pair you've chosen in Live to your headphones. If your interface features direct monitoring, you won't need to activate Cue monitoring in Live for recording, but you will to hear the recording play back.

Under control

While Live is, naturally, compatible with any MIDI keyboard and/or control surface you care to point at it, it's also by far the most well-supported DAW on the market in terms of bespoke, dedicated controllers designed for clip-triggering, mixing and device control.

Arguably, the best of all for live performance is Ableton's own Push, designed in conjunction with Akai. This impressively solid dark grey slab houses an 8x8 pad matrix that can be used to trigger clips and play virtual instruments (the latter incorporating powerful scale mapping functionality), along with eight multi-function rotary encoders and soft buttons, and a whole host of other performance-orientated features.

Hot on Push's heels, though, comes the Akai APC40 and its slightly disappointing second iteration, the APC40 MkII. This time the central matrix is only 8x5 and can't be used to play instruments, but the APC makes up for it with its omnipresent mixing controls (including level faders), crossfader, Device Control rotaries and superb overall build quality.

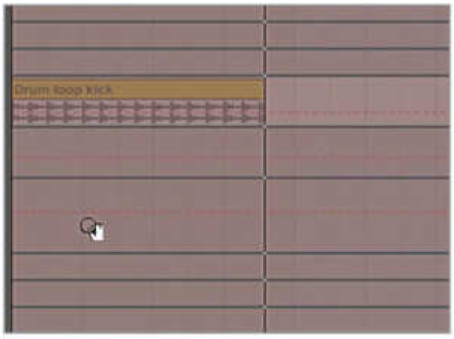
Taking a more modular approach, Novation offer a whole range of Live-centric controllers, each with its own specialisation. The Launchpad S and Launchpad Mini are primarily about clip triggering, while the LaunchKey range adds 16 clip triggers to various controller keyboard configurations, and the Launch Control and Launch Control XL specialise in mixing and device manipulation.

As great as these 'Abletonised' boxes are, though, they're certainly not the be all and end all when it comes to hands-on control of your Live sessions. Like we said, any MIDI controller can do the job, although we would recommend seriously considering portability if you actually intend to take your tunes out of your studio and onto the stage. With that in mind, the compact likes of Korg's nanoKontrol series, Behringer's X-Touch Mini, Akai's MPK mini, and the many creations of German manufacturer Faderfox are well worth checking out.

Push is a great controller for live work, but others exist

> Step by step

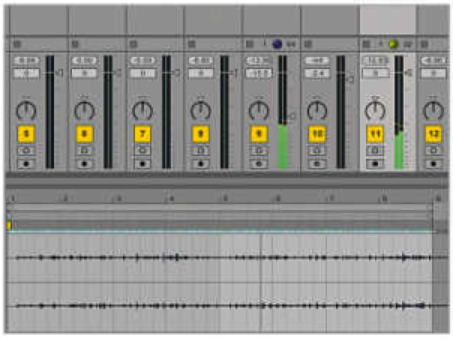
3. Performing the set



1 > We're ready to perform! Delete everything off the Arrange page, including Locators (you *did* make a new version of the project, right?) and make sure **Launch Quantization** is **Enabled** in the Options menu so that clips wait for the next downbeat to play/stop. All of the clips in our session (except the 'Outro 2' Scene) are set to loop, so there's no danger of the music stopping.

2 > We activate recording in the transport so that everything we do is captured in the Arrange view (for rendering and distributing as a remix, plundering for ideas that could improve the original track, or just reliving the performance), and start with the bare minimum, building up the deconstructed drum and bass loops before dropping back down into the 'Intro' Scene, then moving up to 'Drums In'.

3 > Having rendered all our parts at their intended mix levels from the original arrangement, we don't need to adjust any volume levels here. Now that we've entered the Scene structure proper, we play around with track mutes using the computer's mouse and the buttons on our Push controller.



4 > After a bit of noodling, we drop everything out and fall back to the deconstructed drum loops, which we then use to underpin a lengthy build-up using the other parts. We want to use the Bells part from the Breakdown in this section, but it only fills half the clip, so we make a copy at the bottom of the track and pull back the **Loop End** brace so that it loops constantly.

5 > Using the Spot FX Drum Rack is a simple matter of hitting keys on our MIDI keyboard to trigger crashes and drop sounds. We can also drag effects plugins straight into the session (although we'd probably only do this using Live's own devices, so as not to compromise stability). We've also bunged Beat Repeat on the drums to bring some glitch to our chilled-out section.

6 > We want to try a Filter Delay on the slap bass, so we mute the channel in the main output and audition it through headphones on the Cue monitoring bus first (quickly, so that the rest of the track doesn't go unattended for too long!). When we've got a sound we like, we drop it back into the mix. When we eventually drop it out again, we bypass the effect, which is only meant for this section.

7 > We kept three of our synths running live so we could jam around with them. Live 9 enables device automation to be recorded into clips in the Session view. Copy a LuSH-101 bass clip to the bottom of the track (keeping the Scenes unaffected), arm the track for recording, click the **Session Record** button in the Transport and throw the inserted EQ around. The moves will be recorded.

8 > Finally, the Auto Filter on the Master output gets in on the action for a bit of cheesy filtering. We sweep it using the knob that we assigned on our keyboard in the previous walkthrough. With our jam all done, we can listen back to it and edit it to our heart's content thanks to the recording of it that we captured in the Arrange View.

POWER TIP

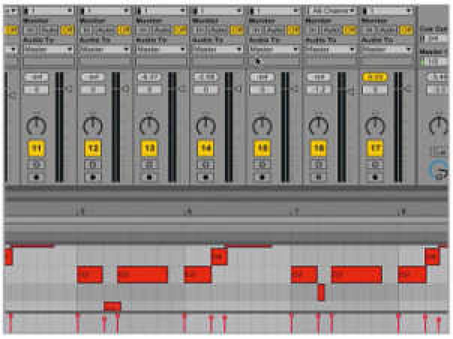
>Mental management

Two important things to keep in mind when improvising with a Live session in front of an audience are the status of your mutes (few things will spoil a drop quite as much as launching a big drum loop or bassline on a channel that you've forgotten to unmute) and edits made to important parts that you might need in their original form later. A perfect example of the latter causing minor issues can be seen and heard in this walkthrough's video at around 7:30!

> Step by step
4. Recording MIDI live into the session

1 > Since we've got our MIDI keyboard hooked up, it would be rude of us to not have a go at recording some MIDI into the session. First, we temporarily unhitch it from the Spot FX Drum Rack by turning that track's **Monitor Off**, and enable **Record Quantization** so that our notes will be perfectly in time.

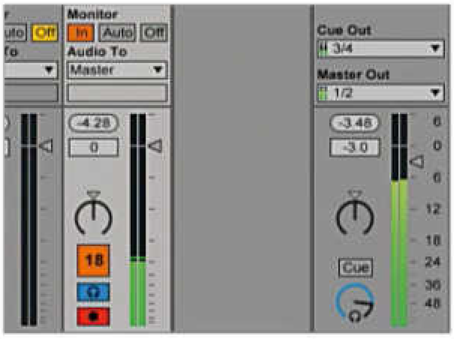
2 > Rather than drag in a new instrument and risk a glitch or, even worse, a full-on crash, we'll use our existing instance of Omnisphere. We **mute** the Omnisphere pad channel, activate **Cue monitoring** (you might have to click the master Solo/Cue button first), set its **Monitor** to **In**, and come up with a part that we like on the keyboard.



3 > With the track armed for recording, we click the **Record** button in an empty clip, record our MIDI part, then click **Record** again to stop recording and start playback of our perfectly looping new clip. When we're ready, we **unmute** the channel to bring the part in. Obviously, the less time this process takes with a track like this, the better!

> Step by step
5. Recording audio live into the session

1 > Let's record audio from a 'real' instrument as part of the performance. Create a new track and set it to the audio input your instrument's plugged into. We mute the track and switch it to the Cue monitor output while we find a good preset on our Korg Wavedrum. Having found it, we unmute the track.



2 > The audience can now hear our live playing; we could do all of this 'silently', but it'd be odd to watch us playing an instrument they can't hear. Arm the track for recording, click the Record button in a clip slot and, when recording starts, play for a few bars. Stop recording afterwards and mute the track again.

3 > As quickly as possible, we shorten the loop within the new clip to one bar (the one that felt the 'best' when playing), then add and edit some Warp markers to get it rigidly quantised, as befits our decidedly electronic track. We could revisit this clip later to extend the warped loop if one bar fails to satisfy.

4 > Finally, we throw on some effects (Live devices only, remember, unless you're madly confident in the solidity of your third-party plugins) to surprise the audience (who will hopefully realise that what they're hearing is a flamboyant new version of the part they just watched and heard us record) and unmute the channel at the next suitable point.

Sooper Looper

While any and all of Live's built-in effects devices are very much on the table when it comes to on-the-fly processing of your jams, Looper in particular stands out for live work, giving the confident performer a simple but effective tool for capturing complex layered instrumental or vocal performances in real time.

Looper is Ableton's take on the looper pedals used by guitarists (and other musicians) to overdub and play back multiple stacked recordings as a live performance. It couldn't be easier to use. Load it onto a track as an effect, click the Multi-Purpose Transport button (or trigger it from a MIDI controller) to start recording, play your instrument and click the button again to stop and initiate looped playback. Then repeat the process to overdub as many further layers as you like without skipping a beat. Once you have a loop you like, you can either leave it running in Looper or throw it into an audio clip slot using the Drag Me! button.

Important note: be absolutely sure that the Song control field isn't set to Start & Stop Song, otherwise stopping playback in Looper will bring your entire set to a grinding and embarrassing halt.

Tips

MIDI pads – such as Arturia's Beatstep – may be a better choice than keys for triggering non-musical material in your set

ILL EFFECTS

We've mentioned it in the main text, but it bears repeating: don't ever do anything during your live set that runs even the slightest risk of making Live hiccup or crash! Apart from the obvious suggestion that you do a few dummy runs before firing it off at an unsuspecting audience, if you do find the nerve to drag any plugin instruments or effects into the project during playback, either stick with Live's own devices (which are bomb-proof in our experience), or those that you know to be completely stable. Also, stick with low-latency effects – preferably Live's own – to avoid timing issues and glitches when loading, and don't touch any latency-related parameters such as lookahead.

LIVE AND DIRECT

If your audio interface offers it (which it almost certainly will), do be sure to take advantage of direct monitoring when recording audio clips on the fly. You won't be able to monitor with plugin effects in place, but the audience will hear them, and unless your processing chain is particularly bonkers, you should be able to successfully do your thing without them. Zero-latency monitoring when playing live is – most of the time – far more important than the player being able to hear the final front-of-house sound. If you just can't handle hearing yourself totally dry, though, some interfaces include 'core' DSP effects (reverb, delay, compression, EQ, etc), which can be applied to the direct monitoring signal – better than nothing!

GOING THROUGH THE MOTIONS

When performing live on a computer, you'll want to do everything you can to maximise your visual stage presence. No, no, no… put the giant foam pointy finger and 'mad for it' jester's hat down – that's not what we mean. We're referring to such things as exciting-looking instruments (electronic drums, keytars, etc) and motion control systems. These days, the latter – which enable you to control anything capable of receiving MIDI CCs by moving your hands in front of a sensor or camera, or using an accelerometer-equipped transmitter to beam movement data back to a receiver attached to your computer – are more affordable than ever. If you've got an iPad or iPhone, IK Multimedia's iRing is well worth a look; and if you haven't, there's the Hot Hand USB, amongst other dedicated systems. MIDI learn some effects controls and wave 'em like you just don't care!

IK Multimedia's iRing is a fine option if you like to get your hands in the air on stage

CRASH PAD

If you don't intend to perform any live melodic parts within your set but do need the ability to trigger spot FX, you might as well go for a set of trigger pads rather than a MIDI controller keyboard. Not only are pads more satisfying than keys to strike out at for such purposes, but with fewer of them in place (eight or 16, usually) than the minimum 25 keys of a keyboard, keeping track of them should be easier, too. Native Instruments' Maschine Mikro, Akai's MPC Element, Arturia's Beatstep and IK Multimedia's iRig Pads are all fine and affordable examples.

FOLLOW ACTIONS

One of Live's most interesting and powerful live performance features is Follow Actions. We've not covered this nifty system in the main text because it's aimed more at generative sound installations than on-the-fly remixing of existing arrangements, but if you want to add an element of chance-based clip triggering to your gig, it's well worth experimenting with. By assigning a Follow Action to a clip, you instruct it to trigger another clip from within the same Session View column when it finishes playing, options including Previous, Next, First, Last and Any (random). Two Follow Actions can be assigned at once, with the relative likelihood of either one triggering being adjustable, and possibilities for their usage within an otherwise 'planned' set could include dropping in randomly selected drum fill variations or spot FX at regular intervals, or rearranging chord progressions on the fly without compromising their overall tonality.

DJ PRODUCER

Fancy taking your live Live sets to the next level? How about mixing a series of them, two at a time, DJ-style? 'Simply' get two laptops, both running Live, and slave one to the other so that they run in sync (via dedicated MIDI interfaces or ports on MIDI controllers); hook their outputs (either built-in or via audio interfaces) up to a DJ mixer; load a separate Session into each and crossfade between your real-time remixes as desired! The headphones situation could get a bit confusing, so we'd suggest leaving out any live auditioning and recording of new parts – unless you want to make things *really* complicated by working in a headphone mixer. For a rather less hectic option, substitute the second laptop for your phone or tablet and just alternate your mix between Live jams and straight playback of regular tracks (giving you time to load up and prepare the next Live set). cm

Norwegian producer Lindstrøm's complex live rig has to be dependable, and so should yours be. If you're going to integrate other parts with your live setup, make sure they won't cause any latency, dropouts, crashes or conflicts

Creative SIDECHAINING

DOWNLOAD
Get the tutorial files to follow along on your PC/Mac at vault.computermusic.co.uk

Sidestep modulation and automation, and give your sounds a fresh jolt by making one jump to the beat of another

> The more experienced producers reading this tome will be familiar with the concept of sidechaining, the most common use of which is for sidechain compression, one of the most widely deployed mixing techniques in electronic music production. With sidechain compression, rather than the compressor processing an input signal based on that signal's own dynamic characteristics, it processes it based on the dynamic characteristics of another signal – otherwise known as the 'key' input or sidechain. This key signal might be audible or silent, depending on whether it's an existing element of the mix or generated solely to trigger the compressor.

Sidechaining isn't only used for compression, though, and when you also consider that sidechaining is, in essence, a form of audio-controlled modulation, the creative possibilities begin to multiply. Compared to the relatively inflexible shapes generated by LFOs and envelopes, the modulation characteristics of audio-based sidechaining are practically limitless. Just one example would be generating precise rhythmic modulation using audio signals that change over time – modulating the modulators, if you will. And of course, any sound source is fair game; in fact, the less obvious the sidechain source, potentially the more interesting the results. You've probably sidechained a synth pad or bassline with a kick drum to impart a rhythmic feel, but why not try controlling an effects processor using your own voice in real time? Or how about synthesising a precise sound designed specifically to control an effect in exactly the way you want?

With this tutorial, we aim to expand your creative sidechaining know-how, and we're going to do it using the 'hidden' parameters of Ableton Live's built-in effects. These walkthroughs will take you far beyond the typical volume pumping associated with sidechain compression as we experiment with filtering, delay, reverb and more to create complex and unique sounds.

Of course, Live's own Compressor isn't the only option available to the sidechain experimentalist – there are endless alternative sidechain-enabled dynamics processing plugins out there to which you can adapt and apply the techniques we're about to show you.

Sidechain, meet synth

Generally, any change over time in a synth sound will come from its built-in modulation systems or an effects plugin. Many effects include LFOs for modulation, while most, if not all, of your synths will feature at least the standard array of envelopes and LFO waveforms – sine, square, sawtooth, etc. A wide range of results can be squeezed out of these standard modulators, but using them exclusively means you're limiting yourself to the same tools that everyone else is also using. Wouldn't you rather come up with your own more interesting, less regimented modulations?

There are several advantages to be gained by processing synths with sidechained effects instead of (or in tandem with) their own modulators. Effects offer great diversity in terms of dynamic sound shaping and are capable of delivering results that are often more characterful and, unpredictable, since the keying signal can be any sound you like, rather than a cyclical waveform or an envelope with a fixed number of stages. You could use, for example, your own speech, kick drums, other synths, found sounds, acoustic instrument samples... Your creativity really is the only limit. Not only that, but you can combine sidechain keying signals: a kick drum, say, followed by a vocal snippet, then a guitar, then a synth pulse wave. The loop of the keying signal (if, indeed, it's looped) can be of any length you choose, and you have full control over the exact timing at which the triggers occur, leading to a wealth of interesting and potentially unpredictable rhythmic diversity. And if you use another synth as your sidechain keying signal, you can even modulate that as well – modulating the modulator itself!

Another powerful source of modulation is, of course, Live's automation system. Indeed, recording or drawing parameter modulation as automation offers one major advantage over sidechained modulation, in that it gives a great visual overview of the resulting movements, which is particularly useful when arranging a track, allowing precise builds and drops to be constructed with ease. What's trickier to achieve with automation, though, is the individual character you get by sidechaining one audio signal from another. Consider the complexity of the average audio waveform in contrast with the straight lines and curves that automation offers.

Ultimately, all modulation sources have their pros and cons. Sidechain processing, however, is potentially the most unusual and creative modulator of all.

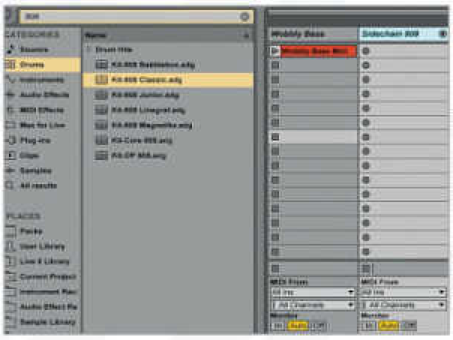
> Step by step

1. Wobbly weirdness with Auto Filter

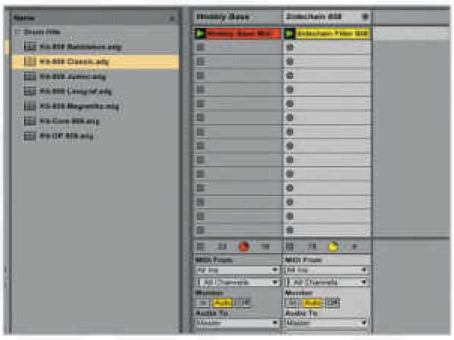
1 > Load Live's **Analog** synth onto a MIDI track in a new project. For the sake of organisation, rename the track 'Wobbly Bass' by right-clicking the track title bar and selecting **Rename** from the dropdown menu.

2 > We're going to program a wide, deep detuned bass, so we start by lowering Osc2's **Octave** setting to **-1**, turning on **Uni**son and dialling in **45%** of **Detune**. Now drag and drop the MIDI clip **Wobbly Bass Midi.alc** (found in the Tutorial Files folder) into a Session View clip slot on the Wobbly Bass track.

3 > Insert an **Auto Filter** plugin after the Analog synth. Change it to **band-pass mode** and set the **cutoff** to **3.13kHz** and the **Q** to **1.18**. Reveal Auto Filter's Sidechain panel by clicking the disclosure button on the left (that's the downwards-pointing triangle), then click the **Sidechain** button to activate it.



4 > We need an audio source to feed into the Auto Filter sidechain. Select the **Drums** category in the browser and type '808' into the search bar to find the **808 Classic** kit (or any basic 808 kit). Drag it to the right of your Wobble Bass track to create a new MIDI track. Rename this track 'Sidechain 808'.

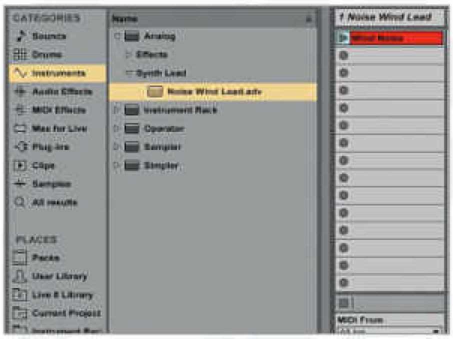


5 > Drag and drop the **Sidechain Filter 808** MIDI clip (provided in **Tutorial Files**) onto the Sidechain 808 track – this is a simple drum pattern. Return to the Auto Filter Sidechain panel and in the **Audio from** dropdown menu, select the **Sidechain 808** track.

6 > It's important to mute the Sidechain 808 channel (by clicking its yellow channel number to turn it grey), as we only want the sound to trigger the sidechain, rather than actually be heard. Play both clips, and raise the sidechain **Gain** to **21.5dB** and the **Dry/Wet** balance to **84.1%**. Set the **Envelope** depth to **-18.1** with **3.9ms Attack** and **60.4ms Release**.

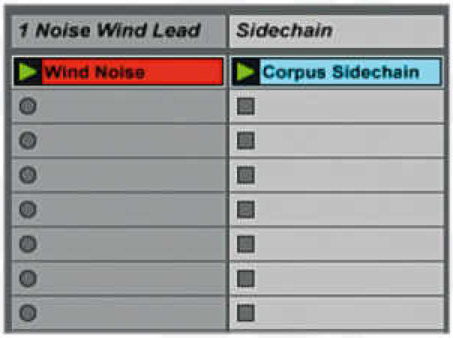
> Step by step

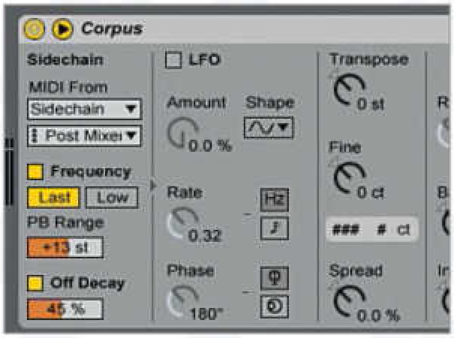
2. Trippy techno with Corpus' MIDI sidechain



1 > Let's create a crazy bouncing techno-esque tone using a synth running through Live's Corpus effect – we'll use the latter's MIDI sidechain input to control it. Find the Analog patch **Noise Wind Lead** in Live's Browser and drag it into the Session View to create a new MIDI track. Next, drop the **Wind Noise** clip in the **Tutorial Files** folder onto the track.

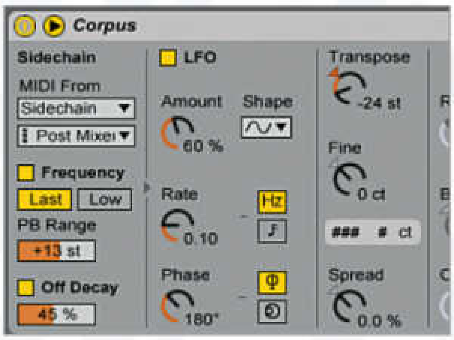
2 > Insert the Corpus effect onto the channel, then reveal its Sidechain panel by clicking the disclosure arrow in the effect's title bar. Corpus' sidechain accepts a MIDI input, not an audio one, so create a new MIDI track – we'll be using this as our sidechain source.





3 > Rename the MIDI track 'Sidechain' and drop the **Corpus Sidechain** MIDI clip provided onto one of its clip slots. It's good general practise to name tracks appropriately, and this is particularly true when sidechain sources are involved, since it's not always obvious at a glance where they're routed to or what their function is.

4 > On the Noise Wind Lead track, select the newly created **Sidechain** track in Corpus' **MIDI From** menu. Activate the sidechain's **Frequency** module and select **Last**. Set the **PB** (pitchbend) **Range** to **+13**, activate **Off Decay** and set the amount to **45%**.



5 > Let's dial in a suitably wild setup in Corpus. First, change the Resonance Type from **Beam** to **Tube**, then set the **Decay** to **27.0 s**, **Radius** to **60%**, **Transpose** to **-24** and **Dry/Wet** mix to **100%** for maximum effect.

6 > Finally, let's give our sidechained bass sound a trippy edge by wobbling the pitch of Corpus with an LFO. Activate the **LFO** and set the **Amount** to **60%**. Slow the **Rate** down **0.10** and you're done! Play the clips together and hear how the MIDI notes trigger Corpus via its sidechain – try altering the MIDI pattern to see how it changes the results.

Chain your groove

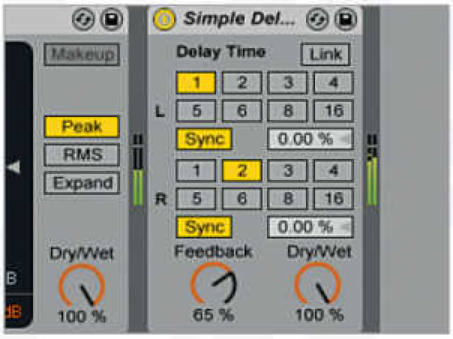
When using audio clips as your sidechaining source, their placement on the track is of paramount importance. This is because the peaks and troughs in the waveform directly affect the timing and depth of the effect, so you can't necessarily just throw any old chunk of audio in and hope for the best – your keying signal absolutely has to be shaped and positioned with the rhythm of the effect you're trying to achieve in mind. Whether that means warping, editing or even re-recording the audio part in order to get it as tight as it needs to be, this is a part of the process that demands your full attention.

While the positioning of the sidechain keying audio is an essential consideration, the placement of the gaps in between clips can be equally important, depending on the effect you're after. It's the appearance and disappearance of the keying signal that triggers and cuts off the effect, defining the groove that it generates. If an audio signal is constantly present at the sidechain input, the effect will always be triggered to some extent, potentially resulting in a "lower contrast" sound that covers less range than one where the sidechain goes from totally inactive to fully active.

> Step by step 3. Making a creative sidechained delay

1 > Let's spice up a typical delay with sidechaining. Create a new MIDI track with an Operator synth on it and adjust the **Level** of Oscillator B to **-20dB**. Activate the **Fixed** button on Oscillator B to make it a multiple of Oscillator A. Set the **Multi**plier to **0.1** and the **Freq**uency to **303Hz**. Drop **Operator arp** MIDI clip (from **Tutorial Files**) onto the track and loop it.

2 > Insert **Compressor** followed by **Simple Delay** on the A Return channel. Open Compressor's Sidechain by clicking the triangular button and activate the **Sidechain**. Set the **Gain** to **5dB**, **Threshold** to **-infdB** and **Ratio** to **4:1**. Now, whenever there is a signal presennt at the sidechain, the compressor will 'shut down' the signal being processed.



3 > In the Simple Delay, set the **Delay Time** for the **L**eft channel to **1** and the **R**ight channel to **2**. Turn **Feedback** up to **65%** and **Dry/Wet** mix to **100%**. We're ready for our sidechain source now, so on a new audio track, place a percussive sample – we're using Clap-707 from Live's library – and rename it 'Sidechain Clap'.

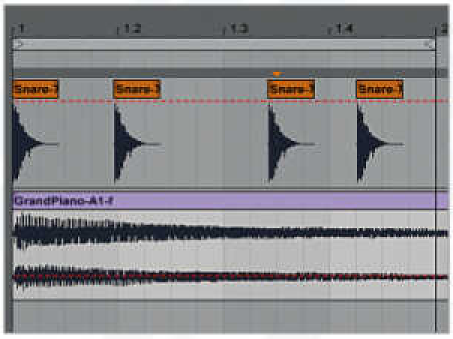
4 > Change the sidechain input on the Compressor to the **Sidechain Clap** track and mute the clap track itself. Turn the A Return **Send** up to **0** on the Operator track to hear the effect. Experiment with adding duplicates of the clap sample in order to control the sidechain delay rhythmically and make it dance!

Simply sidechains

With version 9, Ableton Live introduced sidechain functionality for more devices than ever before. For example, the Gate effect could now be triggered via an external audio signal, giving much more creative control than previous versions did.

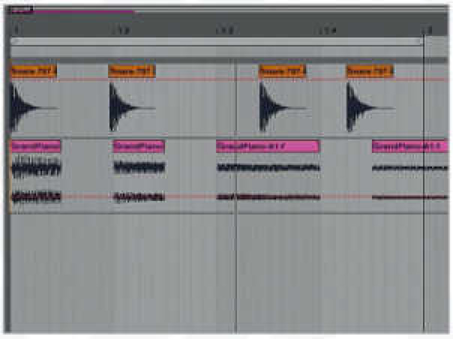
Live 9 also introduced the awesome sidechain-toting Glue Compressor (a reworking of Cytomic's The Glue) and reworked the regular Compressor to give better visual feedback. The Multiband Dynamics plugin, meanwhile, boasts the ability to sidechain-compress individual frequency bands – powerful stuff!

> Step by step 4. Using a sidechain to control a reverb effect



1 > Lay a simple snare sample out four times within a bar, as shown, and loop playback around that bar. Rename the track 'Snare Loop'. Then create an audio channel named 'Sidechain Piano' – we're using a piano sample we grabbed from Live's library, but any sample of a sustained note will do.

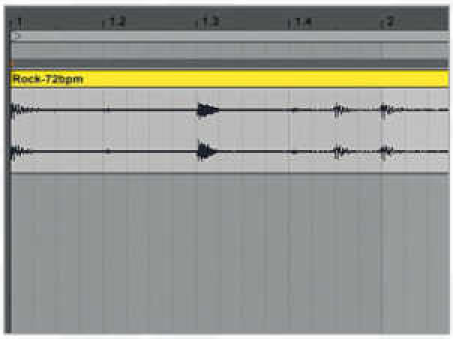
2 > On the A Return channel, add a Compressor followed by a Reverb. Set the Compressor sidechain up as we did in the previous tutorial, but this time keyed from the **Sidechain Piano** track.



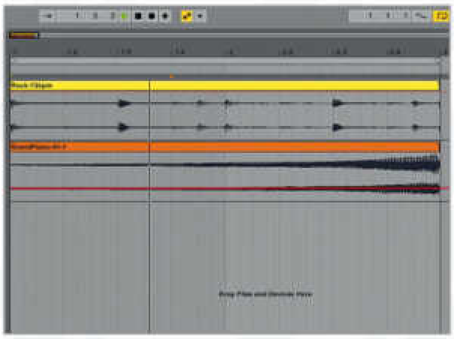
3 > We've supplied Compressor and Reverb presets in the **Tutorial Files** – load them up now. Raise the **Send** amount from Snare Loop all the way up to **0**. Chop bits of the piano loop out by deleting small segments of the waveform – the reverb will jump up between the gaps due to the action of our sidechained compression.

> Step by step

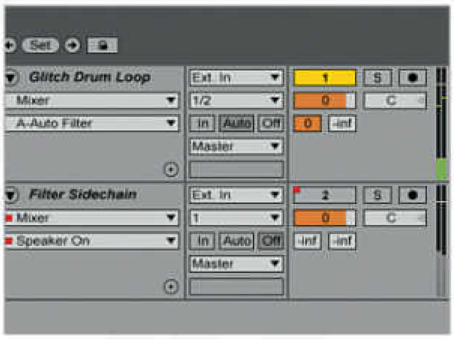
5. Experimental super-sidechain



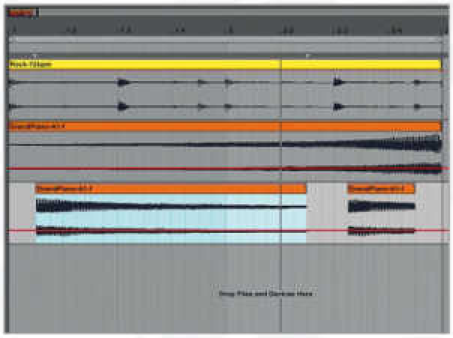
1 > In this final spectacular sidechain setup, we'll be taking all we've learnt so far and applying it to create results similar to 'effects sequencer' plugins like Glitch 2 and Effectrix, using the sidechain method to control the results. We'll start out transforming a typical drum loop into crazy glitched-out experimental drums. Drag a sampled drum loop into the arrange view to get going.



2 > We only want two bars of this loop so we'll shorten it. We'll then right-click on the loop and select the **Loop Selection** option in the dropdown menu. To start with a sidechain filter effect, take the single-note piano sample used in the last tutorial – or any gently decaying musical note sample of your choice – and drop it onto a track below the drum loop. Shorten this sample to 2 bars and reverse it.

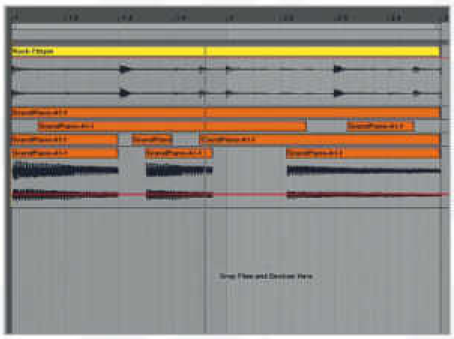


3 > Name this piano track 'Filter Sidechain'. Now add an **Auto Filter** on the return track, load the preset in the Tutorial Files folder and set its sidechain up to trigger from the **Filter Sidechain** audio. Don't forget to ensure the **Send level** for this send channel (A) is raised to **0** (maximum) on the drum loop channel, and that the output volume is muted on the sidechain channel.

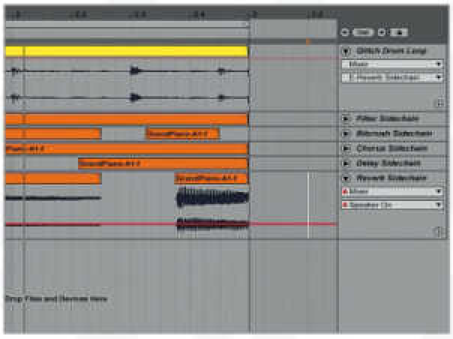


4 > Next we're going to do the same thing with the **Redux** bitcrusher effect – create a new audio track for the same piano sample. Instead of reversing it, we'll resize and duplicate it, cutting out parts of the clip so the sidechain opens on the first kick and the second snare of the glitch drum loop. Name the track 'Bitcrush Sidechain', mute it, and assign it as the key input to Redux.

5 > Now we'll add a different flavour of degradation using some extreme Chorus settings. Create a new return channel and drop the provided Chorus Sidechain preset onto it. Create another sidechain audio channel naming it 'Chorus Sidechain' and assign it as the key input to the compressor before the chorus. Notice how this sidechain again triggers on different drum hits.



6 > Now it's time for some dubbed-out delay to add to this already grubby loop by repeating the process in the last step. Create audio gaps in the sidechain channel, as shown. Using the delay sidechain preset, create a Delay Sidechain return channel and assign it as the key input for the compressor that feeds into the Delay.



7 > To soften the second snare and give it some space, replicate the sidechain setup process, only this time using the **Sidechain Reverb** preset from **Tutorial Files** and spacing the reverb sidechain audio trigger channel so that there is a gap in the audio on the second snare at position 2.3.

8 > Next we'll add some further detuned mayhem with the Flanger. This time we aren't going to create a sidechain for the Flanger – just create another return channel and insert the Flanger preset before turning up its Send amount from the drum track to full. Now you can sequence the effects just by placing snippets of audio visually on the tracks – much more immediate than automation!

9 > To enhance this drum loop harmonically, we're going to add some tube distortion. Place the **Dynamic Tube** plugin directly on the glitch drum audio channel and increase its output to **5dB**. To push out the harmonics, turn the **Drive** to **10dB** and crank the **Envelope** to **295%**. Turn down the master if it's clipping and you're done! **cm**

MAX FOR LIVE ESSENTIALS

Part of Live 9 Suite, Max For Live is an incredible device construction kit that comes with a colourful supporting cast of pre-built instruments and effects…

> When Ableton and Cycling '74 first announced that they were partnering up to bring the latter's powerful device-building environment to the former's hugely popular DAW, our imaginations ran wild. We envisioned a whole new App Store-style sub-market opening up, enabling amateur instrument and effects designers to make their realised ideas available to their peers, heralding a torrent of new toys for us all to take advantage of.

Sadly, though, that didn't really happen. We don't know how well (or not) Max For Live sold after its launch in 2009, but we'd hazard a guess that it wasn't a huge hit. Being only available as a rather costly add-on, its value to those who just wanted to use its included devices, rather than actually make their own, was always a bit questionable. Those devices have always been great, but arguably wouldn't have been worth the asking price on their own.

However, all of that changed with the release of Ableton Live 9! Max For Live (based on Max 6) is now included in the bargainous Suite edition of Live 9 – a brilliant move on Ableton's part that immediately boosted the user base considerably. Non-Suite users can still buy it separately, of course, but the point is that lots of people now have M4L appearing in their Live Browser without having to actually make any kind of buying decision. And even if those users never open up the M4L Patcher editor, they can take advantage of the numerous devices available in the free Max For Live Essentials and Pluggo For Live Packs, downloaded from Ableton's website.

Over the next four pages, we'll make some interesting, characterful sounds using the varied array of contraptions that comprise Max For Live Essentials. If you like what you see and hear, be sure to check out Pluggo For Live's numerous devices, too…

> Step by step

1. Programming beats with DrumSynth

1 > Max For Live Essentials includes a collection of instruments known as DrumSynth – 13 drum synthesis modules that can be loaded individually or as a kit within a Drum Rack. We start by loading a Drum Rack onto a MIDI track, loading **DrumSynth Kick** from the **Max For Live Essentials** Live Pack into the C1 cell and triggering it on every beat with MIDI.

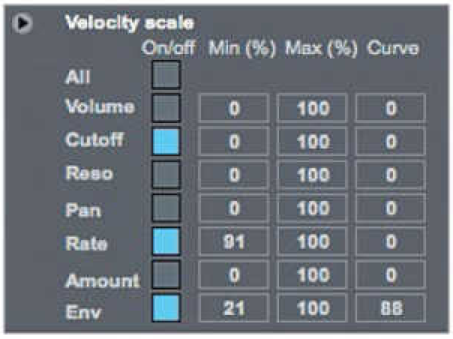
2 > The default sound is pretty dull, so let's liven it up a little. First, we lower the **Pitch** to **E0** and raise the **Decay** to **39** for more depth and sustain. Then we turn the **Drive** up to **89** to give it some bite. The Noise section houses a noise oscillator, complete with its own envelope and multimode filter. We set it as above to add just a touch of sculpted white noise to the kick.

3 > We need a snare, so we load the **DrumSynth Snare** module into the next slot of our Drum Rack, **C♯1**, and draw in the obvious MIDI notes on beats 2 and 4. The most important areas of the Snare module are the Snap, Pitch and Noise sections, which we set as shown to make the sound much more pleasant. We also get rid of any pitch modulation by setting the **Pitch Env** and **LFO** controls to zero.

4 > Hi-hats next, and for these we use two **DrumSynth HH** modules – one for the closed hats and the other for open. For our closed hats, we raise the **Pitch** and **Decay** a little, then darken the sound by lowering the **Color Tone** control and raising the **Resonance**. For the open hats, we up the **Tone** and **Decay** to stretch and brighten the sound. The pattern plays on the offbeats.

5 > DrumSynth's **Clave** module features a Repeater section, which makes it easy to generate 'bounces'. After programming a suitably Latin-esque MIDI pattern for it, we drop the **Cutoff** and **Resonance** to make it sound more realistic. Cranking up the **Repeater Amount** introduces repeats, and upping the **Rate** takes it from a very fast delay to, effectively, a single tone. The **Env** control sets the length of the repeats.

6 > The Clap module actually offers two sounds: Clap, obviously, and Brush – a filtered noise generator with an envelope. We turn the Brush volume up and raise the **Decay** and **Smear** a bit – Decay extends the sound, Smear dials in a fast delay. We also switch the filter to **High-pass**, raising the **Cutoff** to keep the Brush tone top-heavy. For the main Clap tone, we raise the **Decay**, **Cutoff** and **Resonance** to add bite.



7 > The little blue arrow button at the right of each module opens the Advanced window, in which each and every parameter can be made velocity-sensitive. The **Min** and **Max** values set the depth of modulation, while **Curve** adjusts the response curve. We set the Clave module's **Cutoff**, **Rate** and **Env** to respond to velocity, then adjust the note velocities in our MIDI clip to create some movement.

8 > As an alternative to MIDI programming, Max For Live includes a nifty drum sequencer called **InstantHaus**. This triggers Kick, Snare, Hi-Hats and Perc sounds on your choice of MIDI notes, using a range of preset patterns chosen with the Pattern dials. **Swing** and **Hi/Low** velocity levels give a certain degree of rhythmic control. Pretty much everything can also be randomised!

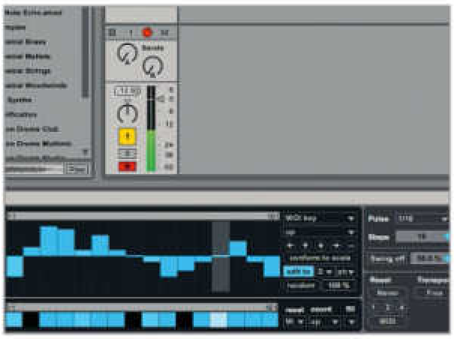
POWER TIP

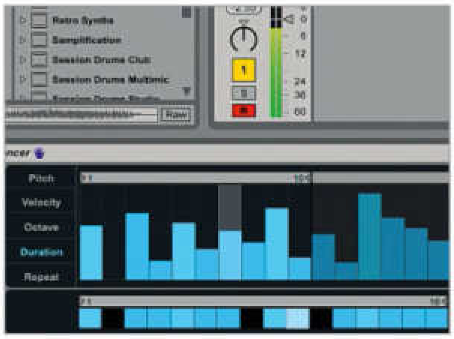
>Random factor

With Velocity on offer as a mod source in all of DrumSynth's modules, dialling tonal expression into your drum tracks is simple and rewarding. For instant weirdness, why not throw in some random velocity variation? Live's Velocity MIDI Effect lets you specify the range of coverage, amongst other things. In **DrumSynth Random.wav**, we've thrown in all of the remaining DrumSynth modules, triggered by a MIDI clip running alongside InstantHaus, each with all velocity modulation targets activated and a Velocity MIDI Effect inserted, with **Random** cranked up. Mad enough for ya?

> Step by step

2. Max Essentials MIDI Effects

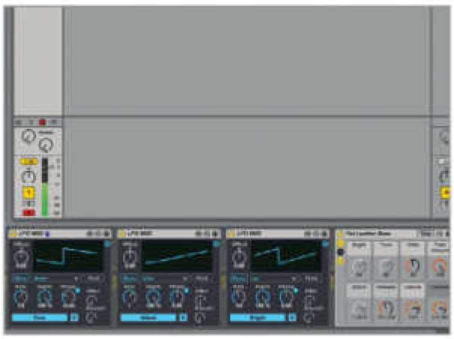




1 > Max For Live is just as capable of creating MIDI devices as it is synths and audio effects. Mono Sequencer from Max Essentials is a nifty step sequencer that lends itself particularly well to basslines and lead synth parts. We load up Live's **Fat Leather Bass** Instrument Rack and insert Mono Sequencer, which begins playing when we start playback in the main transport. (**MIDI Effects 1.wav**)

2 > We stay at the default 16 steps and program a sequence of notes – Note On/Off is in the bottom lane, pitch is in the one above. Steps can be snapped to a specific scale and mode – we go for **D Phrygian**. Playing or programming MIDI notes transposes the whole sequence, with C3 being the default position.

3 > There are additional sequences for controlling Velocity, Octave, Duration and Repeat (a machine gun effect, not hugely useful for our bass sound), each with its own loop length, which opens up all sorts of pseudo-polyrhythmic possibilities. We set up control sequencers for **Velocity** and **Duration** – 13 steps long for the former, ten for the latter. (**MIDI Effects 3.wav**)



4 > Up to 12 patterns can be stored at once, switched between by the **Pattern** knob, top left, which is modulated via automation or by switching the **MIDI Mode** – by default, set to Transpose – to the **Patterns** setting. Now, instead of transposing the sequence, our MIDI clip switches patterns. Helpfully, our pitch changes are automatically retained as Transpose automation.

5 > Let's add further movement to our sequences. Among Max For Live's most useful Essentials are the LFO and Envelope devices, which can be quickly mapped to almost anything in Live, just by clicking the **Map** button, then the target parameter. We set up three **LFO MIDI** devices to modulate the **Tone**, **Attack** and **Bright** Macro knobs in our Instrument Rack, as shown.

6 > Envelope generates a MIDI-triggered ADSR envelope that can be used to modulate just about any Live parameter. Let's use it to control the LFOs we just added. We set up two Envelopes, one modulating the waveshape of our **Bright** LFO, and the other modulating the mod depth of the first. (Live currently seems to have trouble retaining LFO Sync settings in project files, it's worth noting.)

Further control

In the walkthrough on this page, we're looking at Mono Sequencer, LFO MIDI and Envelope. We'll get into Note Delay later on, but there are a few other 'control' devices in Max For Live Essentials. The effortless mapping assignment system is a consistent feature across all of them: simply click the Map button for the modulation control source, then the target parameter.

Min% and Max% parameters govern the range of parameter movement for that control. Map, say, MultiMap's Input dial to a filter's cutoff with Min% at 40 and Max% at 90, and the dial will move the cutoff between those two amounts. By setting Min% higher than Max%, lowering the control raises the target parameter. Adding the filter's resonance to our example, with Min% at 90 and Max% at 40, raising the Input dial will lower the cutoff and boost the resonance.

The aforementioned Multimap enables mapping of up to eight parameters (remember, these can be *any* controls in Live, including those of third party plugins), each with its own Min% and Max%, to a single Input control for simultaneous manipulation.

Expression Control maps five standard MIDI controllers (Velocity, ModWheel, Pitchbend, Aftertouch and Keytrack) to five targets. The input from each controller can be a linear or logarithmic curve, and Attack and Release times can be set from 0-1000ms. It seems odd that the full range of MIDI CCs isn't available, but nonetheless, Expression Control makes setting up keyboard modulation in Live absurdly easy.

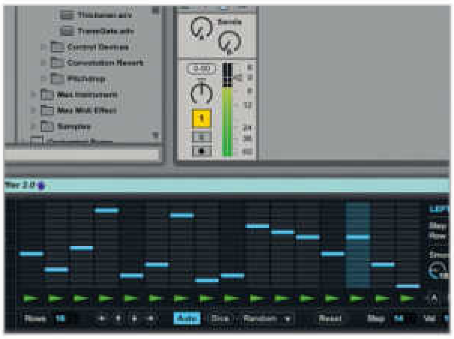
XY Pad allows you to map a pair of parameters to the X and Y axes of a mousable control area. Each assignment has its own Min% and Max%, as well as an adjustable response curve. X and Y can themselves be modulated, of course – endless chucks are to be had assigning an LFO device to each one.

Device Randomizer is our favourite, though. This mad little goblin maps to an entire device (click Map, then the target device's title bar) in order to randomise as much or as little of it as you like. You can set how often a new random value is sent to each parameter freely from 0-40Hz or synced to host tempo, with a Smoothing control setting the interpolation anywhere between totally smooth and fully stepped. There's even a modulation oscillator with which to apply FM to the randomisation! Lunacy! Individual parameters in the target device can be connected and disconnected in the Edit page, and if you just want to send single 'bursts' of randomisation, rather than a constant stream, that's also an option.

> Step by step

3. Buffer Shuffler 2 and Convolution Reverb

1 > The Buffer Shuffler 2 effect enables slicing and rearranging of loops, as well as sequencing of its various built-in effects. We start by loading the bounced loop from our DrumSynth tutorial (**Buffer Shuffler 1.wav**) and setting Buffer Shuffler's **Length** to **2 bars** and **Steps** to **16**. Before Buffer Shuffler can do its thing, we have to play it the full loop so that it can record and store it.



2 > We have two bars to play around with, at eighth-note resolution. The loop is divided into 16 slices, represented by the vertical positions of the trigger steps, and any slice can be triggered on any step of the sequence. For instant results, hit the **Dice** button at the bottom, which randomly rearranges the slice triggers. To re-dice the loop with each cycle, activate **Auto**. (**Buffer Shuffler 2.wav**)

3 > We deactivate Auto (we don't want *that* much chaos!), program in a deliberate pattern (**Buffer Shuffler 3a.wav**) and set **Smooth** to **0%** to keep the transients as sharp as possible. Now we can play around with the other effects. **Stutter** enables glitchy, Beat Repeat-style repeats to be applied, the step height determining the speed. (**Buffer Shuffler 3b.wav**)

4 > Finally, we draw in sequences for the **Pitch** (frequency), **Shift** and **Pan** parameters, and activate some of the **Slice Reverse** buttons in the slice sequencer. Using the **Pattern** buttons on the right, incidentally, you can switch between ten Buffer Shuffler 2 'states', each one a complete sequenced setup of its own.

5 > We mix the dry loop back in in parallel by setting the **Dry/Wet** mix to about **50/50**. To add further flavour, we turn to Max For Live Essentials' other flagship effect, the new **Convolution Reverb Pro**. This powerful impulse response reverb enables separate IRs to be used for the early reflections and tail, as demonstrated by the very odd **Buffer Shuffler 5.wav**.

6 > Convolution Reverb Pro features a set of shaping controls with which to sculpt your reverb. There's a 3-band Parametric EQ, a Modulator section, Damping and more. We use the **Damp**, **Shape** and **EQ** tabs to rein in our patch and give it a darker vibe.

Pluggo For Live

Although Cycling '74 are best known for Max/MSP and Max For Live, they also saw success in the early noughties with their highly regarded bundle of esoteric instruments and effects, Pluggo. Comprising over 100 devices, ranging from the niche but useful to the overtly bizarre (but also useful!), these wilfully weird gadgets became a hit with producers of electronica and IDM – and rightly so. It's very good news indeed, then, that they've been reborn for the Ableton generation in the shape of the Pluggo For Live Pack, free for owners of Max For Live.

Although you don't get the full Pluggo collection with Pluggo For Live, you do get seven instruments and 40 effects – frankly, since there was a fair bit of chaff amongst the full Pluggo wheat, this whittling down could actually be seen as a good thing.

While the instruments do include a straightforward analogue synth (Laverne), it's the likes of Additive Heaven, Big Ben Bell and Analogue Drums that make Pluggo For Live worth the, er... free download when it comes to sound generation. ShepardTones is another highlight, generating the continuous ascending or descending tone of the so-named technique that always makes a great base layer for risers and fallers.

Pluggo For Live's real strength lies in its superb line-up of effects, though. From self-modulating filterbanks, to crazy delays and distortions, to the legendary Squirrel Parade (which defies description – suffice to say that 'animal modeling' is involved and Harry Hill isn't), there's an 'academic' feel throughout, encouraging a spirit of experimentation.

Like everything else in Max For Live, lush, warm, 'expensive' analogue sounds aren't the order of the day with Pluggo For Live. These are quirky, interesting devices with real grit and character, adding up to a truly timeless collection. If you haven't already, go get it from Ableton's Live Packs page.

Pluggo's 100+ devices quickly gained a cult following, and the best are now available for Live 9

> Step by step

4. Other Max Essentials effects

1 > Let's look at the remaining Max For Live Essentials effects. We start by programming a Mono Sequencer riff triggering the Electric Screamer Instrument Rack and inserting Note Echo in between the two. This is a MIDI delay unit that creates echoes by repeating MIDI notes. **Other Effects 1a.wav** is our dry synth; **Other Effects 1b.wav** is through Note Echo in its default setup.

2 > Note Echo can be synced to host tempo (in which case the eight buttons at the top set the delay length in 16th-notes) or freely set in milliseconds. The delays can also be offset to add swing, pitched up or down by up to an octave, lowered in velocity and even fed back into the input. (**Other Effects 2.wav**)

3 > Envelope Follower is an Audio Effect that uses the amplitude envelope of a sound as a modulation source for anything in Live that you care to modulate. Among the most obvious targets for this is the cutoff frequency of a filter, so we point ours at our Instrument Rack's **Filter Freq** macro to give an auto-wah-style effect.

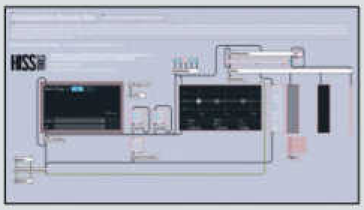
4 > Pitch Drop is a 'slow downer' effect, designed to emulate the 'slowing' sound you get when you switch off a turntable while playing. You can set the length of the drop in milliseconds or synced note values from **1/64** to **8** bars, and two curve shapes are on offer. **Other Effects 4.wav** demonstrates two one-bar drops, the first using the **Soft** shape, the second **Sharp**.

5 > We met the LFO MIDI device earlier, but there's also an audio version that can be used on audio tracks, of course, or placed after an instrument on a MIDI track. In **Other Effects 5.wav,** we've set one up to gently modulate the **Pan** control in the mixer, and followed it with the (non-Pro) **Convolution Reverb**.

POWER TIP

>Acting on impulse

We've used the **Aetherizer 1** IR for our convolution reverb here, taken from Max For Live's impressive bundled library of impulse responses, but we could have made our own using the IR Measurement Device. This brilliant little utility gives you all the controls required to send an impulse or sweep tone out to the device you want to capture and feed the result back into the plugin to generate an IR. It's supremely easy to use, and an included Live Lesson walks you through the process.



The innards of Convolution Reverb Pro – everything's accessible in M4L

Do it yourself

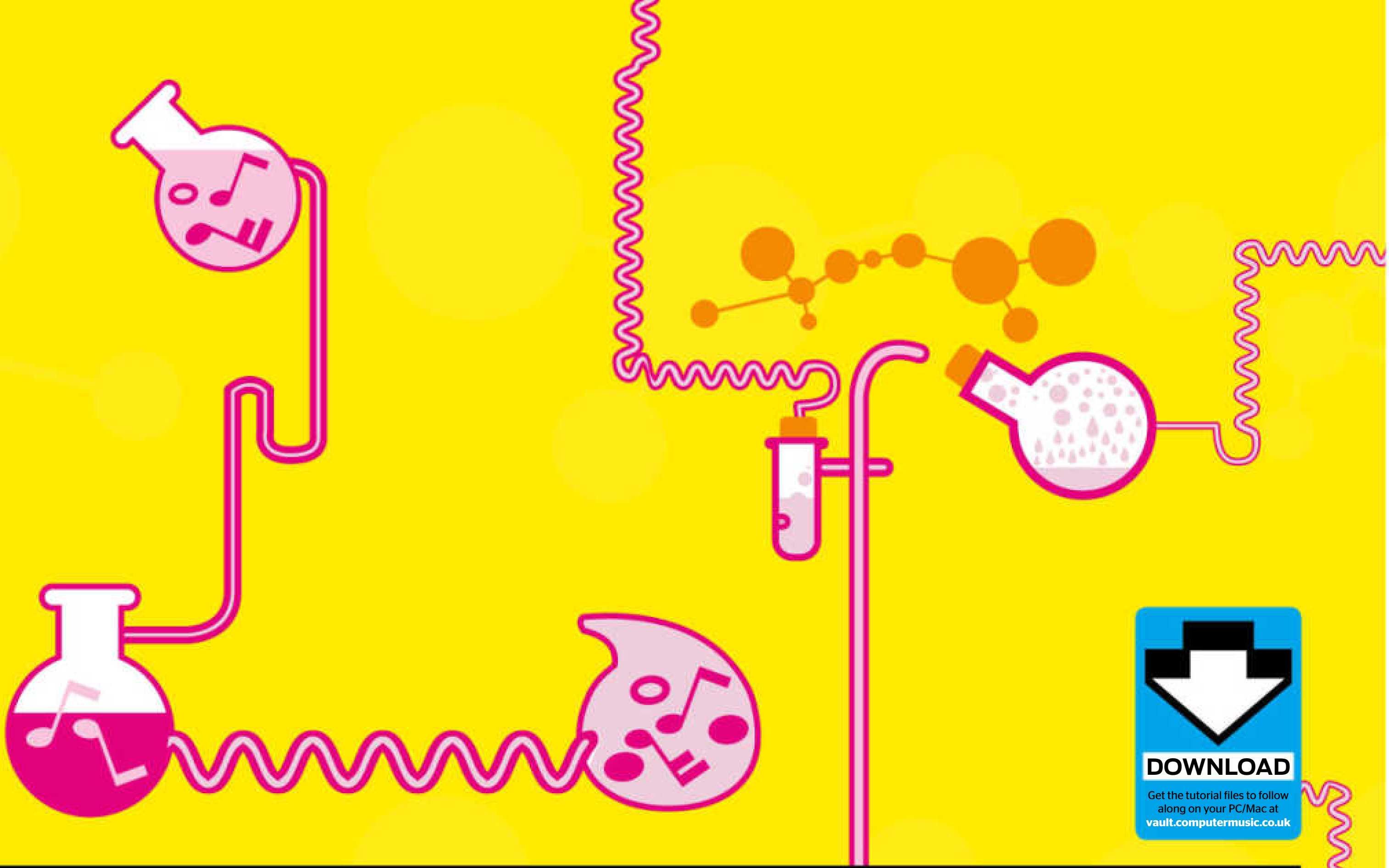
OK, so that's Max For Live Essentials covered, but for many, the real draw of Max For Live will be its power as a platform for building your own instruments, effects and controllers. We certainly wouldn't be so blasé as to describe M4L's device construction process and workflow as 'easy', but it's a long way short of rocket science, and there are a few helpful resources available to get your started.

Your first port of call should be the help system built into Max For Live itself. This is fully hyperlinked, with every parameter and object described in terms of usage, data fields and controls, and example patches accompanying most of the tutorials, which cover everything from the very basics of object creation and signal routing to UI design, video sequencing and hardware control. Crucially, there's also a series of online videos, accessed from within the help system, to walk you through the processes involved in hooking your devices into Live via the Live API. The tutorial system is a fairly monumental achievement on Cycling '74's part, and although it does inevitably get quite heavy, real effort has been made to keep everything as approachable as humanly possible.

Once you're done with that lot, YouTube should be your next stop. There's plenty of educational M4L material on there in general, but we'd recommend starting with www.youtube.com/user/cycling74com and www.youtube.com/learnmax.

Finally, don't forget that you can always open up any of the devices in Max For Live Essentials and Pluggo For Live to muck about with their innards to your heart's content. And once you're done with those, head over to www.maxforlive.com, where you can download loads of new ones. cm

GET EXPER



DOWNLOAD
Get the tutorial files to follow along on your PC/Mac at **vault.computermusic.co.uk**

RIMENTAL!

Stuck in a musical rut? A shot of experimentalism could be what you need to get inspired and cut loose

> The 'experimental' in experimental music is in the ear of the creator. For a house producer, experimental might mean adding a layer of ornamental sci-fi bling to a four-to-the-floor beat. A folk singer might experiment by writing a tune in Phrygian mode (E F G A B C D, in the key of E) instead of a major or minor key. An already experimental composer might flex his über-experimental muscles by going for extremes: John Cage wrote an organ piece that lasts for 639 years, for example.

But what is the *essence* of experimental music, the central quality that all of its various forms share? For the creator: to (boldly) go where they have never gone before. For the listener: to be transported to an unprecedented soundworld. These are no small tasks! Most of us enjoy high musical literacy, surrounded as we are these days by music of all kinds. We are aficionados who are not easily fooled by appearances; we know that a 4/4 dance tune with a theremin teaser is still a 4/4 dance tune, not something truly experimental.

In this feature, we'll create a composition in Ableton Live that visits uncommon realms of musical expression. We'll work in three phases. In Phase I – Gathering Materials – we'll make a map of where we're headed: what layers the piece will contain, roughly how long it will be, its general form and style. Then we'll create each of the main layers – melody, bass, beat, and chords – ensuring that they all 'play well' together.

In Phase II – Development – we'll draft the form of the piece, its jigsaw of component parts. Then we'll arrange the layers to fit this form. Next up are effects; we'll work layer by layer, adding effects to enhance the sound, taking care not to overdo it and end up with a blob of gooshy effects stew.

In Phase III – Honing – we'll add panning and reverb, optimising each track's fullness and vitality. Then we'll do the final mix of all layers and master the mix to make it sing. OK, enough chit-chat – let's get experimental!

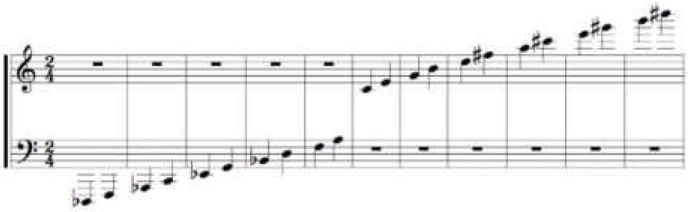
Melody

So, over the next nine pages we'll create a piece of experimental music. First on the agenda is to come up with a game plan. Some composers, particularly those in academia, like to take a top-down approach and construct very detailed plans for their pieces. The benefit of this is that it imbues the composition with a sense of direction and focus – an overarching storyline. The downside is that it's all too easy to mistake an exquisitely crafted plan for the real thing, and to end up fitting uninspired sonic events into inflexible boxes to 'fulfil requirements'.

Other composers, particularly the self-taught, prefer the bottom-up approach of simply diving in and seeing (hearing) what happens. The upside to this is its visceral immediacy – the intimate connection between imagination and actual sound. The downside is that it encourages an undirected approach, which can result in a piece that rambles around rather than telling a coherent and engaging story.

We're going to take the middle path. Top-down-wise, we'll predetermine just a few general things: Our piece will consist of four main layers (melody, bass, beat and chords); it will be arranged into two main sections with a bridge in between; it will exhibit symmetry on multiple levels (pitch, rhythm, form, etc); and it will last about six minutes. That's the right amount of planning to get us started, and just enough to push our track into unusual places that we might not have explored otherwise. Of course, this isn't a recipe or blueprint for experimental music – you are encouraged to come up with your own rules and restrictions, or lack thereof!

Bottom-up-wise, we'll proceed as follows: Compose the pitch scale for the piece, then create first drafts of its four layers: melody, bassline, beat and chord progression. After each layer is done, we'll make sure it plays well with the other layers (rhythmically, harmonically and stylistically). Then we'll develop, refine, mix down and master our track.

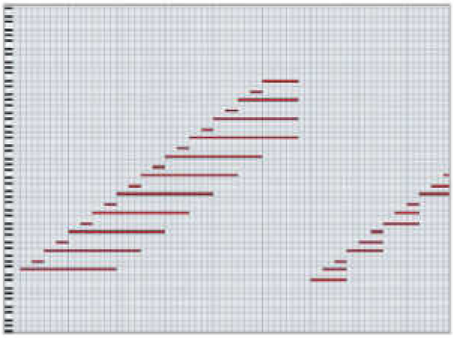


Building the pitch scale for our track will provide a starting point and set the tone for the journey ahead

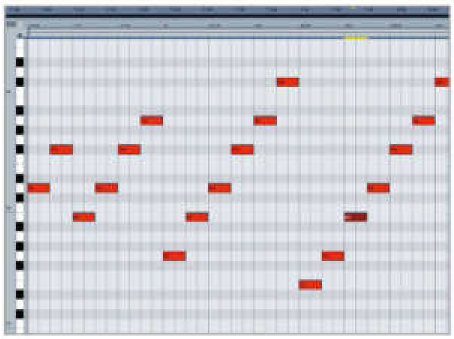
> Step by step

1. Creating a scale and writing a melody

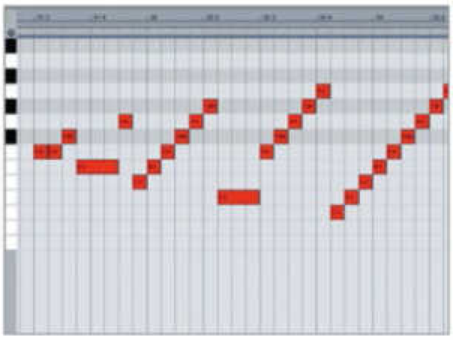
1 > Let's compose us an experimental piece! We start by creating a scale. Unlike conventional scales, which repeat at the octave, our 'experimental' scale repeats at the fifth: **D♭ F A♭ C E♭ G B♭ D F A C E G B D F♯ A C♯ E G♯ B D♯**. It's a variant of the scale used in the great 60s sci-fi TV show, *The Outer Limits*. Have a listen. (Audio: **1 Scale.mp3**)



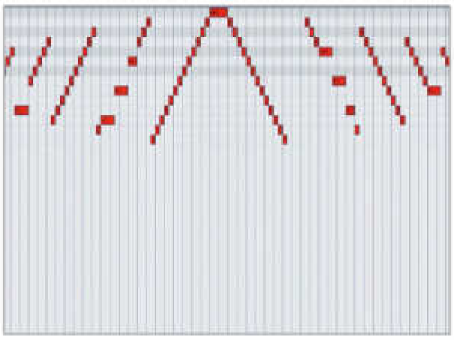
2 > Harmonically, there are two very cool things about this scale: It's built on two sets of ascending fifths (**D♭ A♭ Eb B♭ F C G D A E B** and **F C G D A E B F♯ C♯ G♯ D♯** – think the circle of fifths), and it alternates major/minor triads (**D♭ F A♭** C E♭ G **B♭ D F** A C E **G B D** F♯ A C♯), etc. Sustained notes help you hear these – first the fifths, then the alternating triads. (**2 Scale Fifths and Triads.mp3**)



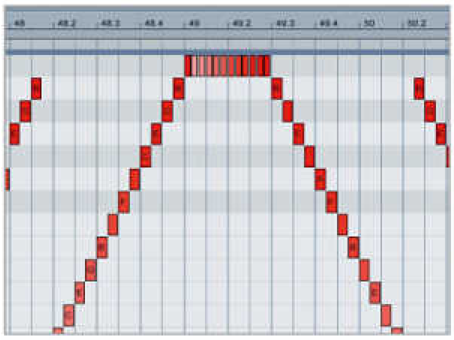
3 > Now let's compose a melodic line with our scale – we'll use a simple rule to create it. The line consists of a series of ascending runs. For the first run, we start with the middle note of the scale (**D4**), then place the next (higher) note right after it, giving us **D4 F♯4**. For the next run, we duplicate the first one but precede it with the next note down the scale and end it with the next note up: **B3 D4 F♯4 A4**. And so on. (**3 Melody.mp3**)



4 > Our melodic line is perky and quirky – two welcome qualities in an experimental piece. But the succession of unbroken 16th-notes is too mechanical; it needs some irregularity. So we remove nine notes and, to bridge the gaps that this creates, sustain the preceding notes. It's a small change to implement, but it makes the melody sound really different. (**4 Melody Pauses.mp3**)



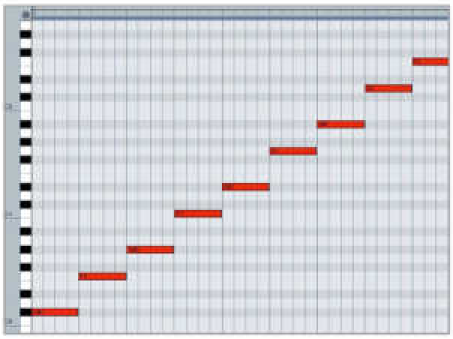
5 > The use of symmetry is a structural technique that you can apply to all musical parameters. We apply it to the melody by treating its final note (**D♯6**) as a pivotal point and mirroring the preceding notes around it in reverse order. It's a strange feeling when this mirroring occurs, like a skewed sonic déjà vu. (**5 Melody Mirrored.mp3**)



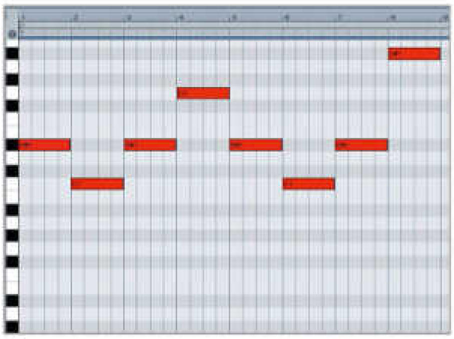
6 > The held **D♯6** pivotal note in the melody is a dead moment – a missed opportunity for sparkle. Let's fix it! Taking inspiration from Frank Zappa's repeated-note marimba writing, we change the **D♯6** quarter-note into a half-note divided into 12 16th-note triplets, and increase the tempo from 120 to **130bpm**. We've also varied the MIDI velocity to add tension and variety. (**6 Melody Sparkle.mp3**)

> Step by step

2. Making the bassline



1 > Next up: bass. What we're looking to create is a slow and steady bassline to act as a foil for the fast and acrobatic melody. In terms of pitch, we already know what notes are available: those contained in the scale defined on the previous page. Here, for reference, are its lowest nine notes. (Audio: **1 Scale low.mp3**)

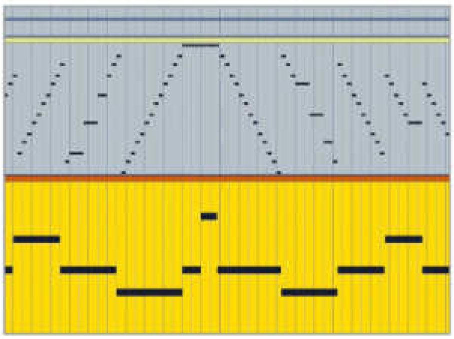


2 > Here's a first stab at the bassline. It might sound too staid for an experimental piece, but even a boundary-thwarting song needs a sense of internal balance. If every layer of the piece was equally frenetic, we could end up with so much high-energy chaos that there'd be no room for the listener to find a way in. (**2 Bass.mp3**)

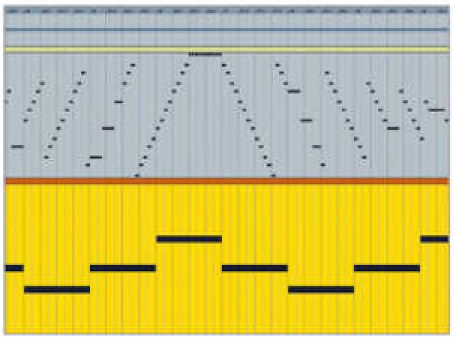
POWER TIP

>Bassic sounds

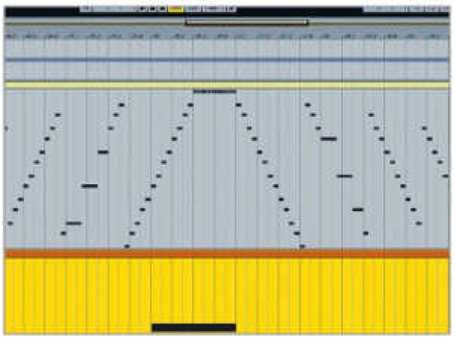
For now, we decide to have the bassline played on an upright acoustic bass patch. The patch's 'pluckiness' (sharp attack, full-bodied sustain, longish decay) add clarity and warmth to the bassline that can help make the piece sound less clinical and more organic. A touch of vibrato might up the warmth. We'll work on the actual sounds some more later on, but for now, we'll stick with this basic patch. Check it out in **3 Bass Acoustic.mp3**.



3 > The next task is to arrange the bassline so that it 'plays well' with the melody. For a first attempt, let's synchronise the start times and durations of the bass notes with the most prominent points of musical energy in the melody – the highest pitches of its ascending sequences. Here's how it sounds. (**4 Bass Melody 1.mp3**)



4 > The bass and melody arrangement we just came up with is nice – off-kilter but with a clear underlying logic. Let's try a very different take and compare. In this version, we place the bassline on steady whole notes, and add an extra note near the end for a little spice. It's a tough call, but we decide to stick with this simpler second version to highlight the melody's complexity. (**5 Bass Melody 2.mp3**)



5 > The bassline is quite high. Since the melody is also quite high, it might be more effective if the bass were lower. Let's take it down by a fifth, which we can 'legally' do, since our scale repeats at the fifth. Nice! The same line a fifth lower works a lot better, with much more gravitas. (**6 Bass lowered.mp3**)

Believe it or not, we were thinking 'cocktail pianist' while crafting our melody

More melody and bass

We decided to begin by composing our piece's main melody (see previous page) because it's the star of the show, spending the most time in the musical spotlight. If you're working in a non-melodic mode – creating a piece of organised noise, for example – think foreground rather than melody. Crafting a good, solid melody/foreground can provide you with an excellent foundation for a successful piece. It's like coming up with a good hook: once you've found one, the other components tend to fall into place.

One of the main things that attracted us to the alternating major/minor triads scale we created for this piece is the sound of playing each successive note in quick ascending and descending motion. To our ear, it's reminiscent of an arpeggio flourish, like something a virtuoso (experimental) cocktail pianist might play in the middle of a cover of *The Shadow of Your Smile*. That's why the melody we came up with is based on ascending and descending scalar runs.

Once the first draft of the melody is done, we like to move onto the bass part. The sound of an intricately dancing melody above a slow and steady bassline is so fundamental to modern music that it's practically a staple of the trade, and just because we're composing an experimental piece doesn't mean we have to forego it. Instead, we can use it… then thwart it! Skewed conventions can sound every bit as out-there as original inventions, and our bass part is as conventional as it gets: a 4/4 sequence of whole notes.

The beat

In most genres of popular music, the beat rules. To quote the Duke: "It don't mean a thing if it ain't got that swing." While we might feel somewhat cramped by this aesthetic dictum, we acknowledge its ubiquity and will endeavour to make a drum beat for our piece. But what kind of beat should we make?

We could stay within the realm of convention and create a beat that adheres to the grid of pulse and meter. Pulse is the steady tick-tock of musical time, the imagined metronome that clicks on each beat – you might nod your head or tap your foot to it. Pulse speed or tempo is measured in beats per minute (bpm). A pulse of 60bpm has one beat every second; a pulse of 80bpm has one beat every 0.75 seconds. To work out the number of seconds between beats at a given tempo, take 60 and divide it by the bpm.

Meter (or time signature) is the number of beats per bar and the value of those beats, notated as X/Y, where X is the number of beats in a bar and Y is the note value of those beats. A 3/4 meter means that a bar consists of three quarter-note beats, for example, while an 8/8 bar contains eight eighth-note beats.

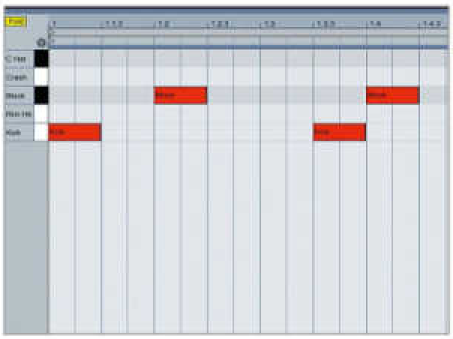
We could stay inside the grid but go outside the norm and fashion irregular meters (9/16), or mixed meters (3/4 + 7/8 + 9/8 + 5/4), or polymeters (3/4 vs. 4/4 vs. 5/4), or irrational meters (11/12 = each bar contains 11 triplets). Tempo-wise, we could go for extremes (a bpm of 0.1 or 960), or wildly varying tempos (slow down from a bpm of 240 to 60 over four bars, or accelerate up to 480 over the next two bars).

We could even go rogue, take the "we don't need no stinkin' grid!" route and create a free-form percussive passage with no pulse or meter, or a beat so utterly skewed that most listeners would hear it as random percussive noise. We'll aim for something in the middle...

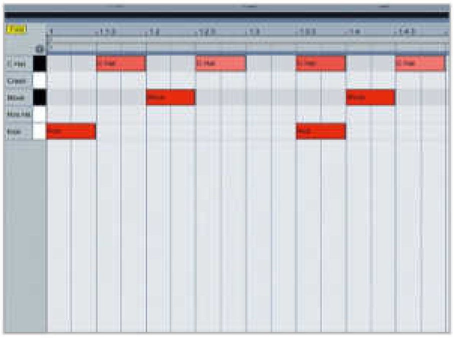
Our drum beat, shown here in good old musical notation, is in 9/8 – which is quite unusual in itself!

> Step by step

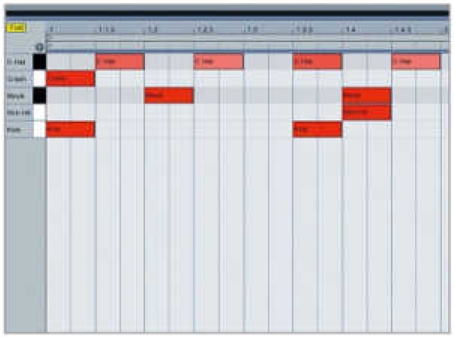
3. Designing an experimental beat



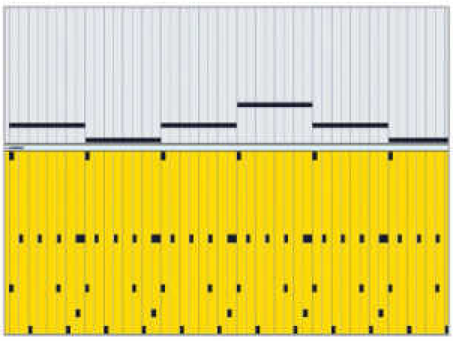
1 > We've got melody and basslines that play well together. What's next on the agenda? The beat, of course! Let's start off nice and simple. We can always throw a wrench in the works later if it sounds too dull. We start with a variant of a conventional 4/4 backbeat: kicks on beat 1 and beat 3.5 (ie, the offbeat between beats 3 and 4) with a 'block' sound on beats 2 and 4. (Audio: **1 Beat.mp3**)



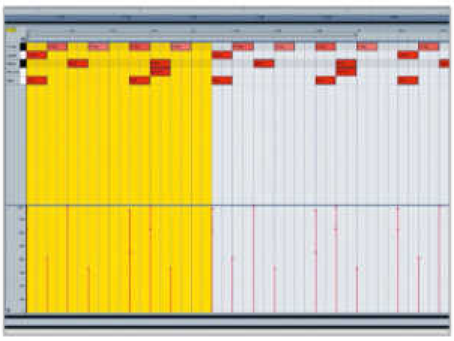
2 > To give the beat forward momentum, we add closed hi-hat hits to every offbeat. The missing beat on 3 works well, inserting a moment of intriguing imbalance into an otherwise predictable rhythm. That hint of start/stop-ness is something we'll pick up on and emphasise later. (**2 Beat.mp3**)



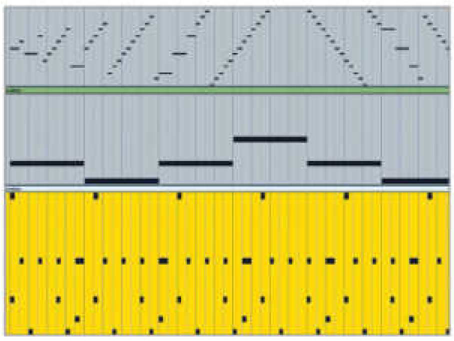
3 > Our beat sounds pretty good, but we're not quite there yet. Let's put a crash cymbal on beat 1 to announce the beginning of the cycle. When the music gets dense and complex, this will help listeners know where things stand rhythmically. As the icing on the cake, we add a rim hit to the offbeat of beat 4. (**3 Beat.mp3**)



4 > Next, we devise a musically compelling (and experimental) way to arrange the beat with the bass and melody. For comparison's sake, we start by syncing the beat (in 4/4) with the bassline (also in 4/4). The result sounds OK, but really, it's too conventional to satisfy our experimental urges... (**4 Beat + bass.mp3**)



5 > Let's try something different: a compellingly imbalanced meter. To do this, we simply add an eighth-note pause to the end of the beat loop, making its meter 9/8 instead of 4/4. The result is, to our ears, quite delightful. It's fascinating how that one extra eighth-note pause changes everything. (**5 Beat + bass.mp3**)



6 > Let's add the melody to the mix – OK, now we're getting somewhere! The music is both attractive and confusing, but in a good way. At first it sounds quite normal and predictable – a nice little trio. But the 9/8 drum beat and oddly contoured melody soon throw a delicious wrench into the works as the three layers start to drift apart rhythmically. (**6 Beat + bass + melody.mp3**)

Chords

We love chord progressions! Done right, they can serve as a gorgeous 'sonic glue' to hold disparate elements of a composition together. If you're working in conventional Western harmony (ie, major/minor scales), chords and chord progressions play a functional role within their scale. So, for example, if you're working in the key (scale) of G major (G A B C D E F♯ G):

- The tonic (I) chord (G major: G B D) functions as 'home' within the key.
- The dominant (V) chord (D major: D F♯ A) leads back home to the tonic.
- The supertonic (II) chord (A minor: A C E), leads to the dominant, which in turn leads home to the tonic.

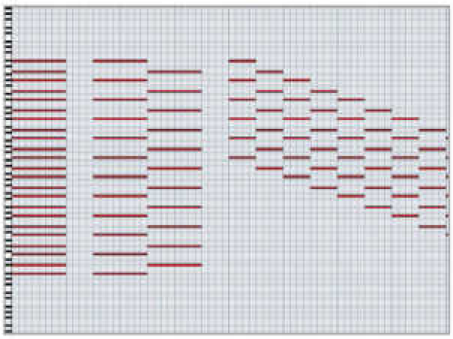
The chord progression from supertonic to dominant to tonic (notated as II-V-I) is very common in Western music.

Because our scale is outside conventional Western tonality, its chords do not function as the chords of major/minor scales. Instead, we must create our own functionality. We design a six-note (two-stacked-triad) chord 'home base': **F3 A3 C4 E4 G4 B4**. Then we make an ascending chordal progression by transposing our six-note home chord up a fifth three times: **C4 E4 G4 B4 D5 F♯5** then **G4 B4 D5 F♯5 A♯5 C♯6** then **D5 F♯5 A♯5 C♯6 E 6 G♯6**. Then we descend by fifths four times all the way down to **E♭2 G2 B♭2 D3 F3 A3**. Finally, we mirror this chordal line going backwards – up four times and down three to get back to the 'home' chord. Because our scale is based on a pattern that repeats at every fifth, all this by-fifths tranposition is perfectly 'legal'.

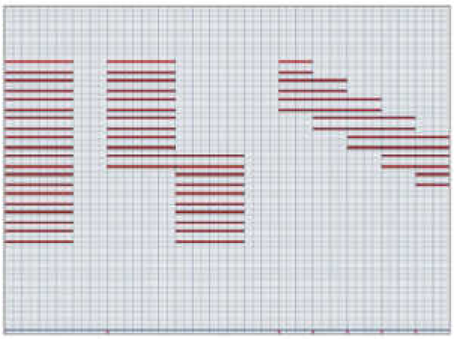
Our chord progression moves six-note chords around by steps of fifths to exploit our scale's fifths-based nature

> Step by step

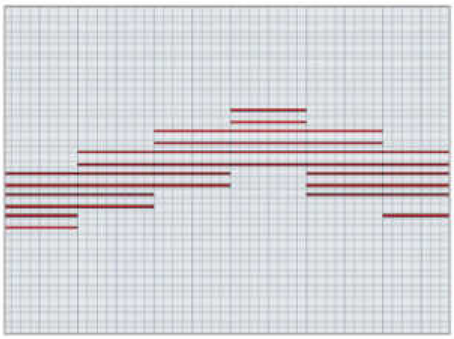
4. Stacked experimental chord sequences



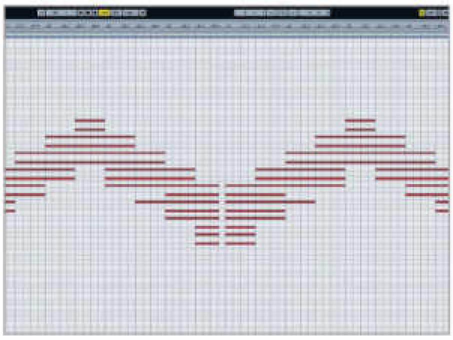
1 > For now, one more layer to our piece will do, and we're going for a chord progression played on a string-ish synth patch. To reveal the possibilities to our musical ear, we play all 23 scale notes in one huge chord, follow it with the scale's two interlocking sets of fifths, and then a cascade of narrower stacked fifths. It's fifths heaven! (Audio: **1 Chords.mp3**)



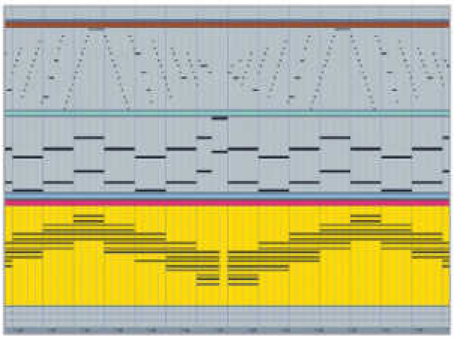
2 > As mentioned earlier, our scale has two prominent harmonic features: the stacked fifths we just heard and alternating major/minor triads. Now let's check out those triads. Here's the entire scale stuffed into one chord, followed by high and low alternating triads, followed by a cascade of stacked triads, which sounds rather sweet. (**2 Chords.mp3**)



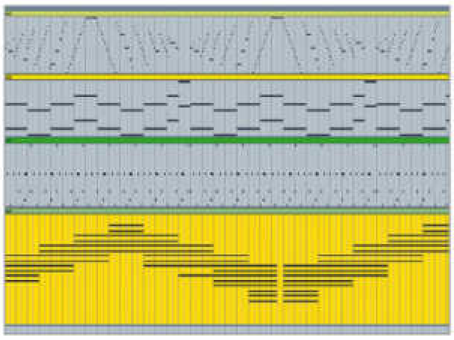
3 > Now that we've got the chordal possibilities fresh in our mind (and ringing in our ears), we can compose our progression. The stacked-fifths chords, though gorgeous, are perhaps too 'bare' sounding for a rich progression. We decide to stick with the alternating-triad chords, arranging them into this sequence of half-notes. (**3 Progression.mp3**)



4 > Self-mirroring is a key element of many works of art. When layers reflect each other, directly or indirectly, it imbues the piece with a sense of internal consistency and strength. We mirror the melody's symmetry by making our chord progression similarly symmetrical around a low central chord. (**4 Progression.mp3**)



5 > We've got two similarly structured layers: a symmetrical melodic line and a symmetrical chord progression. Convention would have us sync them to make their central pivot points coincide, but we're all about thwarting convention! Instead, we play the chords twice as slow as the melody. (**5 Progression.mp3**)

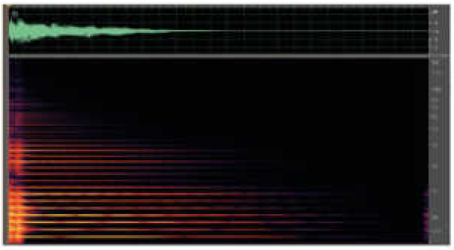


6 > Hmm... our Step 5 solution syncs chord notes with bass notes, resulting in way too much 4/4-ishness. Let's try see if some more mirroring can dig us out of this decidely square hole. Taking our lead from the beat, which is in 9/8, let's put the chord progression in a non-4/4 meter. Here it is in 6/4, against 4/4 (melody and bass) and 9/8 (beat). Better! (**6 Progression.mp3**)

Timbre

Music is sound. It's not enough for a track to be well conceived – it has to *sound* good too. This might seem like a no-brainer, but many pieces fall flat because their sounds are not as engaging as their formal structures. This is especially common in academia, giving rise to paper music that looks great on the page but sounds dull in performance. It's like a well-plotted but boring novel. Non-academic composers fall into the opposite trap, arranging good sounds and sexy hooks in a weak form, like a nicely written but poorly-plotted novel. A good composition needs both: compelling sound and solid structure.

Timbre is the sound of a sound. The timbre of a note is determined by the relative amplitudes of its partials (sine waves at different frequencies), or more accurately, the amplitude envelopes of these partials – how each rises and falls in volume over the duration of the note. If the partials of a note are integer multiples of the fundamental (lowest) frequency, they are called harmonics. Non-integer-multiple frequencies are called inharmonic partials.



It's not simply the frequency content of a sound that matters, but how the harmonics develop over time

Consonant, pitched notes (flute, clarinet, violin, trumpet, etc) are rich in harmonics. Dissonant, unpitched notes (cymbal, snare drum, hi-hat, etc) are rich in inharmonic partials. Our experimental piece uses a mix of consonant and dissonant timbres. The melodic line is played on a consonant vibraphone patch from a soft synth. The bassline is played on an acoustic guitar patch that has a lovely dissonant plucked attack, followed by a consonant sustain and decay. The chord progression is played on a string patch that is a mix of consonant (pitched string) and dissonant (the breathy noise of bowing). And the beat is pretty much completely dissonant, except for the kick and ride cymbal, both of whose fundamental pitches are clearly audible.

> Step by step

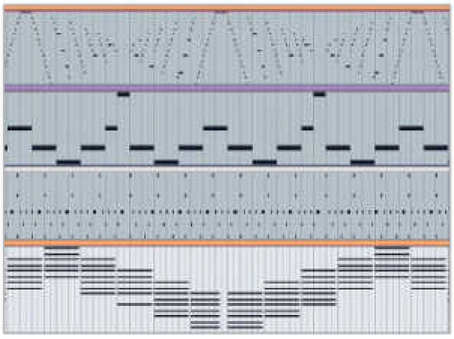
5. Sound selection

1 > Up til now, we've been using blasé General MIDI sounds. This is a great way to work up a rough demo – if it sounds good like this, the music itself must be pretty compelling. Eventually, though, you're going to have to use proper sounds. Opting for Cableguys Curve 2 CM (from **cm** Plugins), we choose the **Vibraphone (CG Edit) LE** patch for the melodic line. (Audio: **1 Melody.mp3**)

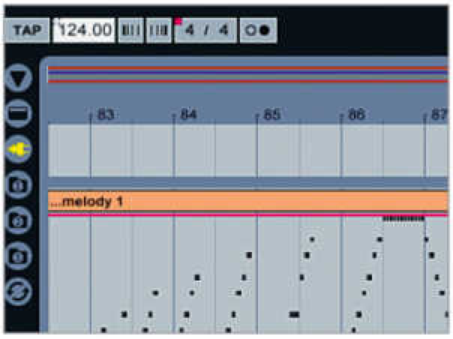
2 > For the bass, we want to preserve the sharp attack and longish decay of the GM acoustic bass patch we've been using. To do this, we use the default **Acoustic Guitar** patch from Plucked String 4, also from the **cm** Plugins collection. It's a good sound, but to make it even juicier, we add a touch of Twin Peaks-ish vibrato using the Vibrato section. (**Audio: Bass.wav**)

3 > On to the drum beat. Once again, we draw upon **cm** Plugins by choosing the **Elektro2** patch from our own CM-505 drum machine. Not unsurprisingly, we have to transpose pitches to make the beat work correctly with the CM-505 MIDI settings. The result is a bit different to our GM-powered effort, but it's eminently usable. (**Beat.wav**)

4 > Finally, for the chord progression we choose the excellent ZebraCM synth. We pick the **SM smooth pad MW** patch for its (digitally) warm string/organ sound. Like most sustained string patches, it works well in all registers, from low to high. Because we have so many notes playing at once, we have to set the **Voices** to **Many** to stop them cutting off with each new chord (**Chords.wav**)



5 > You can hear the full mix using our chosen synths and patches in **Mix.wav**. Not too shabby! The balance isn't perfect, but we'll work on that in the final mix. The important thing is that the layers play well together, supporting rather than fighting each other.



6 > After listening carefully to the new mix, we decide to make a few subtle changes. We reduce the tempo a bit (from 130 to 124) to prevent the melody from feeling rushed, and soften its high notes to reduce shrillness. Also, the chords' long attack times mean they sound a bit late, so we slide their MIDI part forwards about **800ms** to compensate. (**6 Mix.wav**)

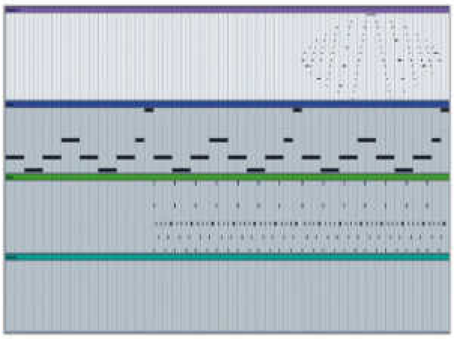
Form

"Form is a cage in which to trap meaning," says an ancient Japanese treatise on aesthetics. Two things stand out in this take on the nature of form. First, the metaphor of form as a cage. There are many different kinds of cage: functional, decorative, simple, complex, an unadorned box, a container so intricate and beautiful that it is in itself a work of art…
The second thing that stands out is the notion that form does not merely arrange parts into a coherent whole, but that it captures and presents meaning.

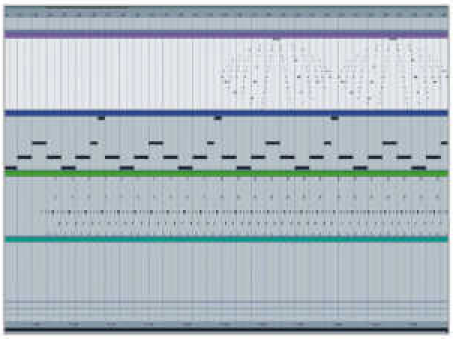
What is the meaning of music? It depends on context: musical meaning, like beauty, is in the ear of the beholder. For a club DJ, music is that which gets the dancefloor moving; for a songwriter, music is a vehicle for self expression; for a scientist, music is the periodic oscillation of air molecules; for a listener, music is sound that evokes emotions and states of being.

What is the meaning of this experimental piece we're creating? From our composer's point of view, the piece's meaning lies in how its four layers of sound interact on all levels (structurally, harmonically, rhythmically, expressively, emotionally) and at all scales from the micro (notes, phrases, passages) to the macro (movements, sections, the storyline of the entire piece). To present this meaning, we'll capture it in a rather simple cage of form. Strange, complex sounds need simple, straightforward forms to act as foils for one another. Too much unrelenting weirdness burns itself out and leaves listeners in the dark.

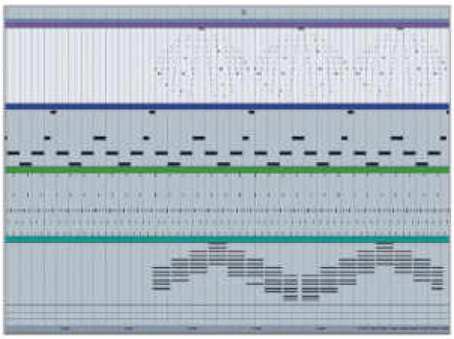
> Step by step 6. Arranging the piece into sections



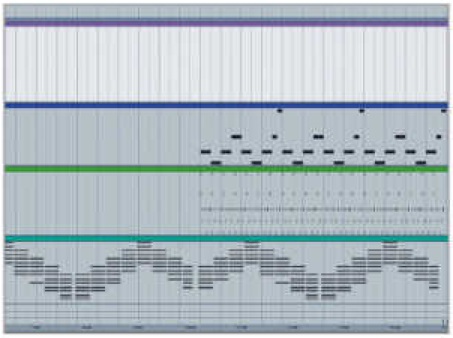
1 > We now have a working passage for our four layers: melody, bass, beat and chords. But it's just 47 seconds long! To create a full-length piece from this, we need to devise a form. We decide to have two main sections with a bridge between. Here's the first of three subsections of section I: (Audio: **1 subsection_1a.wav**)



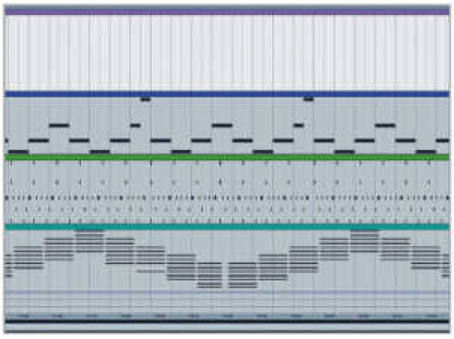
2 > Subsections b and c of section I both grow out of subsection a in a spiral fashion, with each subsection one eight-bar unit longer than the previous. Subsection 1a has 24 4/4 bars: eight solo bass, eight bass + beat and eight bass + melody. Subsection Ib has 32 bars: eight solo bass, eight bass + beat, and 16 bass + beat + melody. (**2 subsection_1b.wav**)



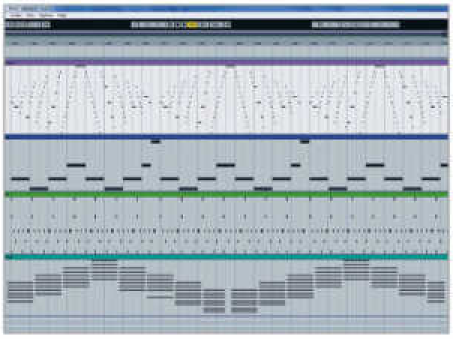
3 > Subsection 1c, which ends section I, adds eight bars to 1b to bring it to 40 bars: eight solo bass, eight bass + beat, and 24 bass + beat + melody. There's a ton of rote repetition in these subsections; we'll fix this later on in the process. Note how the chord progression sneaks its way into the end of 1c. (**3 subsection_1c.wav**)



4 > Next comes the bridge between sections I and II. As foreshadowed in subsection 1c, our chord progression will serve as that bridge, carrying listeners oh-so-gently from the sparse simplicity of section I to the denser complexity of section II. Here's the bridge and the first of three subsections of section II. (**Bridge+IIa.wav**)



5 > Unlike those of section I, the subsections of section II do not increase in duration – they're all 24 bars long. Instead, they increase in intensity, each more dense and complex than the previous. At this point, you won't hear much of that increase – we'll incorporate that later in the process. (**IIb.wav**)



6 > Subsection IIc is the grand finale, bringing together all four main layers (we also intend to work in some supplemental material). Note that the form of the piece (as shown elsewhere on this page), like the melody and chord progression, is symmetrical: Ia Ib Ic – bridge – IIa IIb IIc. More mirroring – yum! (**6 IIc.wav**)

Self-critique

A good artist is a good self critic; not every creation that springs forth from the mind is flawless. There are many traps one can fall into: laziness (taking the easy way out), imitation (copying others and oneself), lapses of aesthetic judgment (that extreme hoover bass really sounded great last night at 2am, but this morning?). Find your compositional Achilles' heels, your cheats, and be ready to pounce on them when they show up in your music. It's critical to listen to your own productions as objectively as possible, so you can improve what's weak and fix what's broken.

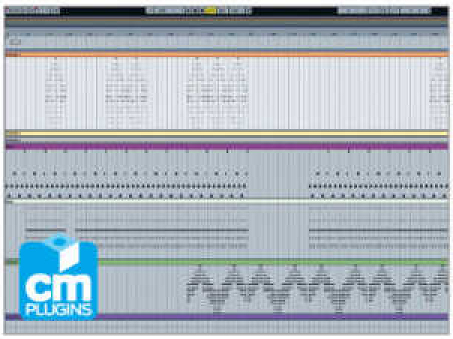
Let's take a good, hard, critical look at our experimental piece in progress. Listening several times to the mix of the arrangement discussed on the previous page reveals strengths and problem areas. Here are the strengths, as we see (or hear) them. Each layer is quite satisfying musically, and the four of them play well together, enhancing rather than detracting from (or competing with) each other. The form (arrangement) has a convincing wholeness about it, both in terms of the relationships among its parts (temporal divisions) and its overarching storyline.

Weaknesses? The arrangement has a mechanical feel, and it sounds a bit skeletal – we've got some good bones but underdeveloped muscles and flesh. The melody sounds thin and rote-repetitive – it needs some variation and sparkle. The bassline is solid, if unremarkable, and it could benefit from some extra twangy vibrato (think Angelo Badalamenti's bassline for the *Twin Peaks* theme). The beat sounds dangerously close to a 9/8 variant of a canned beat on a cheap Casio synth – some additional rhythmic flair would lift it above this. And the chord progression, though quite fine as it is, could sound even more luscious, mysterious and evolving if a second filtered layer were added to it.

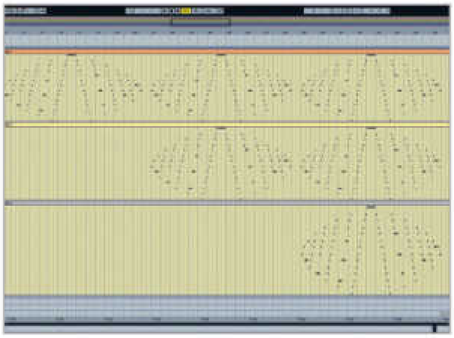
We absolutely don't want our beat sounding like it's being generated by one of these...

> Step by step

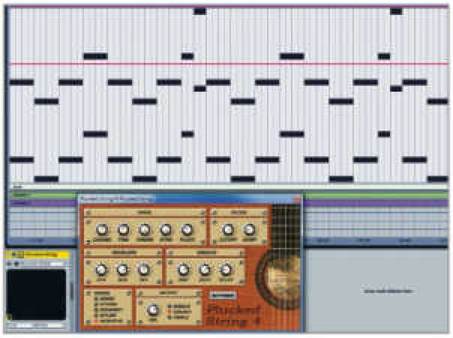
7. Creating variation with arrangement and effects



1 > Now we've got ourselves a full-length piece with four layers, two main sections and a bridge. However, it's still very much in its skeletal state, the equivalent of a detailed outline of a short story. Have a listen, with an ear to the good spots, problem areas and dead zones. (Audio: **Piece_1.wav**)



2 > Now we're into the development phase, and the goal here is to extend and elaborate the material – to add flesh to the skeleton and work the outline into a compelling story. We start by adding two supplemental layers to the melody: the first a fifth lower than the main melodic layer, the second a fifth higher – remember, we used a scale based on fifths for our melody. (**1 melody_dev.wav**)



3 > The bassline is effective; there's not much we need to do to improve it. So we focus on the subtleties of its timbre (sound). We increase the vibrato so that it evolves from relatively mild to positively wobbly. And we double the final passage at the octave so that it doesn't get lost in the mix. (**Bass_dev.wav**)

4 > There's so much we could do with the beat! We could spend hours crafting virtuosic variations and madcap fills, but for the purpose of demonstration, we choose the simpler (yet surprisingly satisfying) solution of KR-Delay CM (from **cm** Plugins) to add a synced feedback delay to the beat. (**Beat_dev.wav**)

5 > The chord progression sounds good as-is – the harmonies, contour, and ZebraCM string/organ-ish patch are all solid. But we can make it even better by adding a second layer – a clone of the first sent through an automated sweep of VPS Philta CM's **Broad Phaserband** patch. (**Chords_dev1.wav**)

6 > We make sure to keep the Philta layer quite far in the background volume-wise, so that it adds just a hint of ethereality and evolving movement to the mix. More than that would be overkill. (**Chords_dev2.wav**)

Three-phase development

Coming up with a good musical idea is only the first step of the compositional process. Even a hook no human ear could resist falling in love with can only go so far in a piece. You can feature the hook and bring it back again and again as a repeating motif, but you cannot build the entire piece solely from that one great hook. This is where development comes into play.

Developing a musical idea is a three-phase procedure. First, you discover the sonic possibilities inherent in the idea. Then you experiment with these possibilities, find out what works and what doesn't. Finally, you present the results of these experiments to your adoring listeners. The good thing is that these principles can be applied to any kind of music, not just the experimental type we're aiming for.

Let's take a more detailed look at each one of these phases.

▸ Phase I: Discovery – Think of the discovery phase as musical brainstorming. As with any brainstorming session, you need to remain open to anything that comes up without passing judgment. Have fun, surprise yourself, go wild! Encourage extreme and even crazy ideas. Experimental music thrives on extreme and crazy, so give it what it needs!

▸ Phase II: Experimentation – Once you've brainstormed a decent set of variants based on your initial musical idea, try them out. Keep one foot firmly in the realm of nonjudgmental exploration and the other in the realm of critical evaluation.

▸ Phase III: Presentation – During this final phase of development, you need to sift through all your experiments, tossing out what doesn't work and keeping only that which does. Strive to be as objective – and merciless! – as possible.

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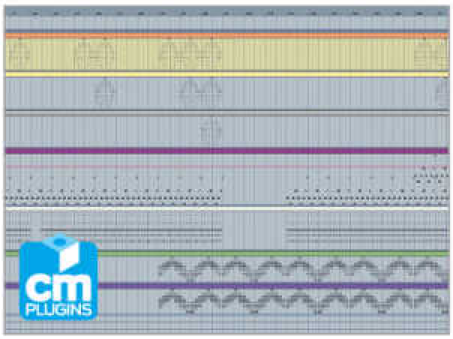
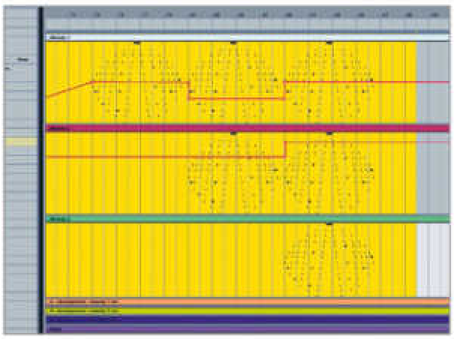
Über-composer Beethoven's Pastoral Symphony is a masterclass in the development of irresistible hooks

> Step by step

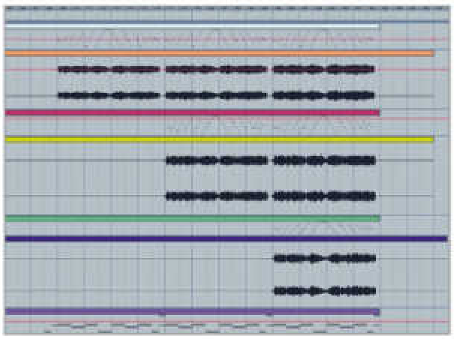
8. Rounding off the parts with panning and EQ



1 > Now we've got a full piece, but it's still not quite there – the volume of the layers needs balancing, the melody needs more development and the tracks need EQing and panning. And most importantly, the piece still sounds mechanical and needs to be made more organic and alive! (Audio: **Piece2.wav**)



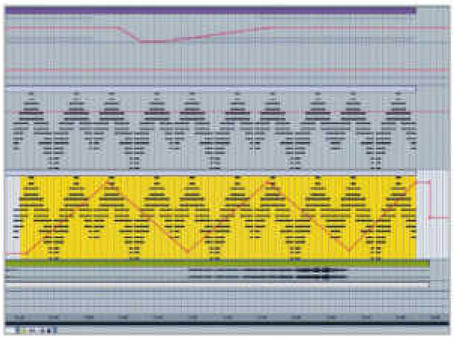
2 > The melodic line is a bit humdrum and lacking in character. To fix this, we start by panning the three melody voices to generate spatial interest – we'll use automation to make it even more exciting. When one voice is playing, we centre it; when two voices are playing, we pan each 40% off-centre; when three voices are playing, we pan 75% off-centre. (**Melody_panning.wav**)



3 > Panning the three melodic voices is a good start, but the melody is still a bit dull. This is especially evident in the grand climax at the end of the piece. To spice things up even more, we create copies of the three voices, add reverb to them, bounce them to audio, then reverse them and mix them with the originals. (**Melody_reversal.wav**)

4 > The bassline sounds quite fine, but some judicious EQ can make it even better. To do this, we turn to the CM-EQUA 87 parametric equalizer, again from **cm** Plugins. To raise the bass while keeping the high-frequency sparkle, we boost by about **5.7 dB** at **100Hz** and **2.5 dB** at **5kHz**.(**Bass_eq.wav**)

5 > The beat could also use some equalisation. This time we avail ourselves of another parametric equaliser, DDMF's awesome IIEQ Pro CM. To accentuate the kicks, we boost by about **3.5dB** at **120Hz**. To up the sizzle by a notch, we use a high-shelf filter of **1.5dB** at **2000Hz**. (**Beat_eq.wav**)



6 > The two-layer chord progression is satisfying, but it would sound a tad more engaging if the filtered layer did a spatial dance around the original. We use pan automation to move the filtered layer back and forth slowly between 80% left to 80% right. (**Chords_panned.wav**)

Final mix and mastering

Let's take stock. We've got four layers of music – melody, bass, beat and chords – that sound good both individually and in combination. We've arranged these layers into a satisfying (if somewhat mechanical) form. We've done some development work that's made the piece more compelling. What's left? We'll trust our most precious musical tools – our ears – to tell us. After listening to the latest version of the piece, we decide to make some final changes.

There are still a few dead zones in the piece – passages of 'same old, same old' that might leave listeners bored. To fix these, we'll add three drones to the mix. The first is a pre-echo of the main melody: a very soft and skewed variant of the melody that foreshadows the actual melody. Adding pre- and post-echoes to a piece can increase its sense of internal consistency and integrity.

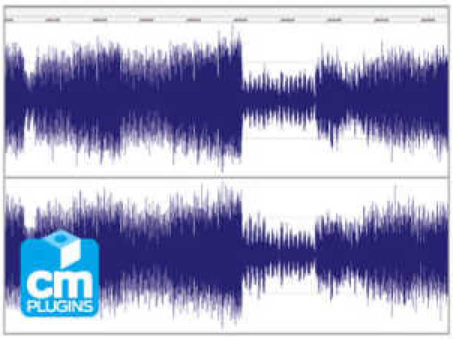
Next we add a sub-bass drone to the piece. There's nothing like a good, strong sustained sub-bass tone a few octaves below middle C to add a sense of more-felt-than-heard gravitas to a mix. It's the sort of thing you can't get away with in most mainstream genres, but here we can really let rip! To balance this 'sub-basso profundo', we add a high-pitched drone chord consisting of seven adjacent notes near the top of our chosen scale. Because this is such a penetrating sound, we use it sparingly.

All that remains is to carry out the final mix and mastering. Before mixing down, we spend some time panning the individual layers and sublayers to spread the sound out across the stereo field. Finally, to cement the sound and finish things off, we customise iZotope Ozone's 'Electro and Dance Rock Master' preset until we achieve a state of experimental sonic bliss.

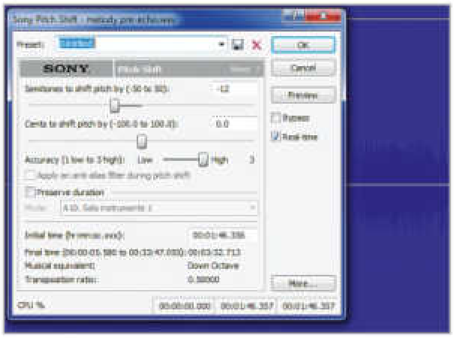
We decide not to get too experimental in the mastering stage, sensibly sticking to iZotope's Ozone 5 plugin

> Step by step

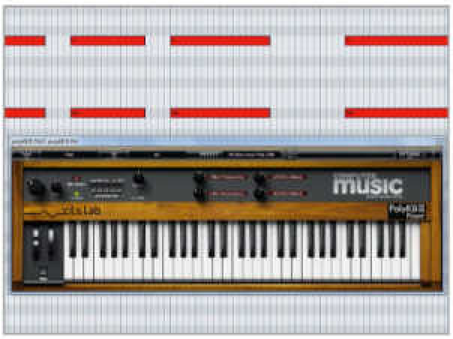
9. Final mixdown and mastering



1 > Now our piece is close to being done! Check it out in **Piece_3.wav**. However, there are still a few less than satisfying moments – places where the piece keeps churning away but nothing new or ear-catchingly different is presented. Let's fix this by adding some drones to the mix. The first one will be based on the main melody, so we render its track out as audio and load into an audio editor.

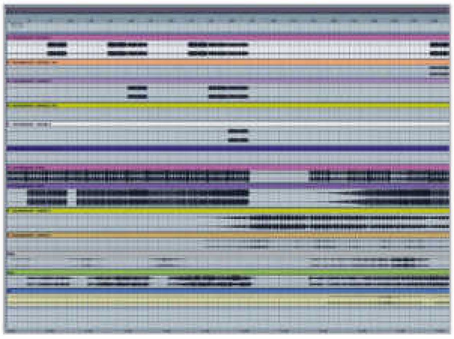


2 > We want the first drone to be a pre-echo of the opening melody. To make it, we take the three-part melodic passage of subsection 1c, slow it to half speed, reverse it, add reverb to it then reverse it again (ie, "unreverse" it). It's an eerie sound and it imbues the piece with a timeless feeling. We save it out as a WAV and import it back into our project. (**Melody_pre-echo.wav**)



3 > Next we create a sub-bass drone to glue things together and create some low-end drama. Adding a sustained sub-bass layer to a piece can give it a more-felt-than-heard sense of awesomeness. We use the **PA Blue Hour Pad** patch of **cm** Plugins' PolyKB II synth to create our drone, two notes a fifth apart (Bb and F). (**Drone_subbass.wav**)

4 > The piece is a little heavy on low and low-mid frequencies, so some additional high-frequency sparkle would be welcome. To that end, we use the **Notch Pad RH** patch of Dune CM to create a high-pitched drone that consists of seven adjacent notes near the top of our scale. (**Drone_high.wav**)



5 > We're all done – time for the final mixdown! We've included two versions of our final mix: one with MIDI parts and virtual instruments, and one with everything rendered to audio, so that you can load it even if you haven't installed all the necessary **cm** Plugins. The project files are downloadable at **vault.computermusic.co.uk**. (**Final mix.wav**)

6 > Just one step left: mastering. The **cm** Plugins collection is geared more towards sound design and mixing, so we'll look elsewhere for a quality mastering solution – iZotope's Ozone fits the bill nicely. We use a modded version of Ozone's **Electro and Dance Rock Master** preset. That's it – our final mix is done and our track is finished! (**6 master.wav**)

TIPS & TRICKS

MISBEHAVE!

When you set out to create an experimental piece, start by getting a sense of the conventional boundaries of the style/genre you're working in. Then you'll have a set of commandments you can blissfully break.

TELL A STORY

All successful works of art tell compelling stories. When you're done with a piece, listen to it objectively and ask yourself: Does this tell an engaging story and take listeners on a satisfying journey? If not, have the chutzpah to go back to the drawing board.

SCALES

The effectiveness of the pitchworld (key, melody, harmony) of a piece depends on its central scale. Choose – or better yet, invent – a great-sounding scale or set of scales, and half the battle is done.

MELODY

Even the most radically experimental music can benefit from beautiful melodic lines. Beauty, in this case, does not necessarily mean: sweet, consonant, lyrical. It can mean: harsh, dissonant, violent.

FOREGROUND/BACKGROUND

Humans parse perceived reality into foreground and background to prevent being overwhelmed by attention-demanding input. Take this into account when you compose an experimental piece: Use volume and/or filtering to place one or two key layers in the foreground and relegate the remaining less-important layers to the background.

RANDOMISATION

Randomisation is an experimentalist's best friend. Using it (wisely!) can bring you to musical places you would never have arrived at otherwise: structurally, melodically, timbrally. The trick is to be able to reject the dozens of unsuccessful results randomisation yields and go for those rare, wonderful-sounding flukes. We think of combining guided randomisation with human intervention (tweaking) as deep compositional collaboration.

TEASING THE GRID

Since drum beats are so deeply ingrained in the expectations of 21st century music listeners, you can have great fun discombobulating them! Food for grid-teasing thought: irregular meters, mixed meters, polymeters, irrational meters, extreme or wildly varying tempos. Or dare to go grid-free: no pulse, no meter, no repetitive beat, just a compelling succession of percussive events.

CLEARING THE PALATE

When you're composing music that consists of (way-)out-there sounds, it's all too easy to overwhelm your ear with sheer audio weirdness. For this reason it's important to take frequent breaks when you're making experimental music, to clear your sonic palate.

© Getty

Our main man, Johann Sebastian Bach, no doubt about to lay the smack down with some mad phat fugues

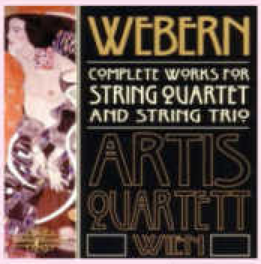
BACH'ING MAD

Ideally, each layer (instrument) of your piece should be compelling both as an individual line and as a member of the ensemble (mix of all layers). Learn from a Bach fugue, whose voices makes musical sense when played individually (solo) and together (polyphony).

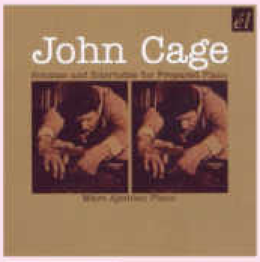
TOP, BOTTOM, MIDDLE

Top-down composers start with a detailed plan. Bottum-uppers jump in and see what happens. Middle of the roaders yo-yo back and forth between the two. Figure out what kind of working method is best for you, and stick with it! **cm**

Recommended listening



Anton Webern
Six Bagatelles for String Quartet
bit.ly/11DXbIA
Listening to experimental music is a great way to get your compositional juices flowing, and this is an atonal masterpiece that any self-respecting experimentalist should know (and love).

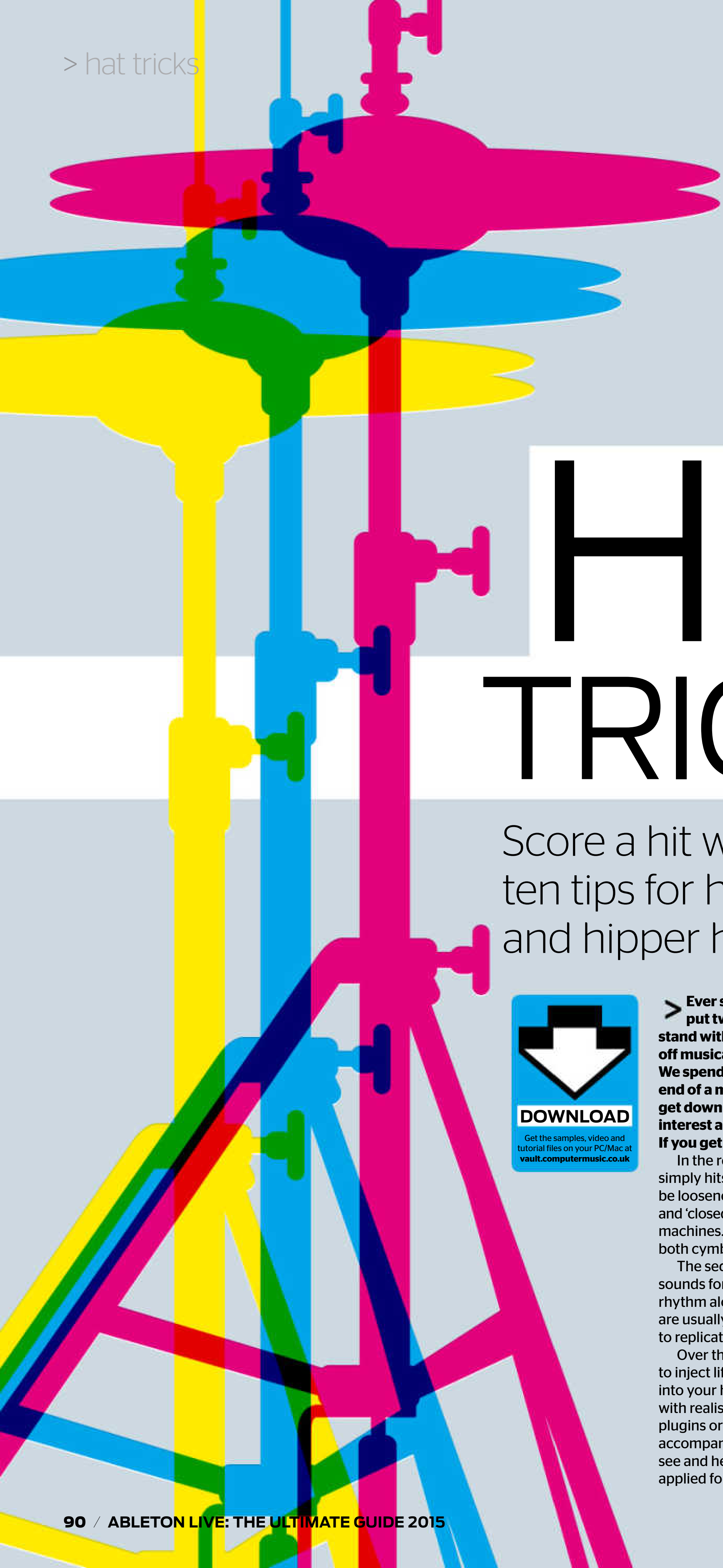


John Cage
Sonatas and Interludes for Prepared Piano
bit.ly/XyA3mp
Cage is known for his conceptual pieces, such as the (in)famous silent 4'33", but the man could actually compose, when he wanted to! These are gorgeous mysterious gems, like hammered sonic dances.

Squarepusher - *My F***ing Sound*
bit.ly/12bCSBH
From the ecstatically virtuosic and genre-expanding IDM album *Go Plastic*, anyone interested in teasing or thwarting the rhythmic grid can learn a huge amount from what happens in this piece with beat, tempo, meter, phrasing, and timbre.

© Getty

Ryoji Ikeda
The Transfinite
bit.ly/WYBtcz
An installation piece for electronic sound and video created by Japanese composer Ryoji Ikeda, what we love about this is how the video and sound complement each other and manage to create a complex sonic-emotional experience with very simple means.



HAT TRICKS

Score a hit with our top ten tips for hotter tops and hipper hats

DOWNLOAD

Get the samples, video and tutorial files on your PC/Mac at **vault.computermusic.co.uk**

> Ever since some bright spark decided to put two opposing cymbals together on a stand with a pedal, hi-hats have been topping off musical grooves in almost every genre. We spend a lot of time talking about the low end of a mix – and sure, that's a tricky thing to get down – but good tops can lend pace, interest and brightness to your tunes. If you get them right, that is…

In the real world, a drummer most often simply hits the hi-hats with a stick. The pedal can be loosened or tightened, leading to the 'open' and 'closed' hat sounds we also find in drum machines. Open hits have a long sustain where both cymbals ring out; closed hits decay quickly.

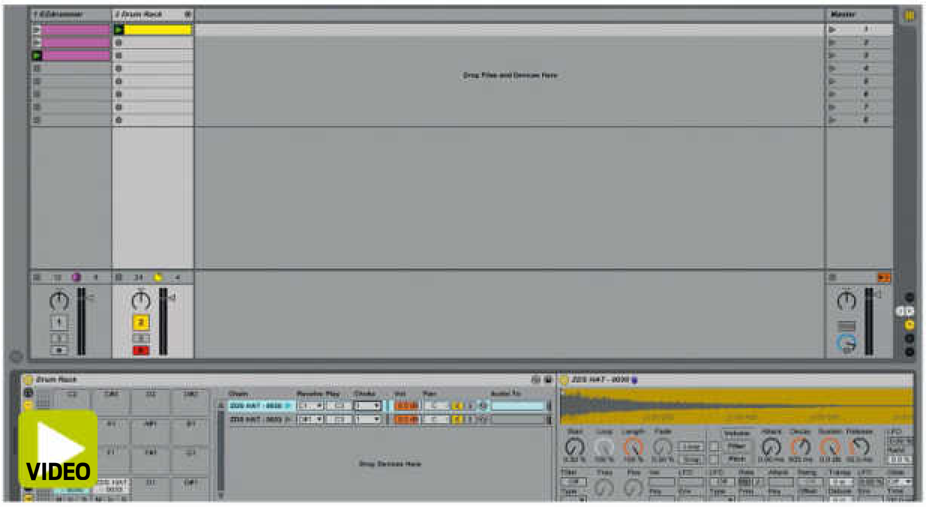
The sequencing of closed and open hi-hat sounds forms a metallic pattern that drives a rhythm along. Synthesised 'electronic' hi-hats are usually created with white noise and filtering to replicate the sound and tone of the hi-hat.

Over the next few pages, you'll find out how to inject life, width, depth, dynamics and realism into your hi-hat parts; whether you're working with realistic drums, electronic drum machine plugins or samples. Every technique features an accompanying video walkthrough, so you can see and hear how each of our hat tricks is applied for yourself. Hold on to your hats!

1. Chopping out a loop's hi-hats

> Before we tackle the nitty gritty of programming hi-hats from scratch, let's first look at how you can use and customise pre-existing hat grooves. Many producers feel that using ready-made drum loops from sample packs is a form of cheating, but it's actually possible to chop up, rearrange and customise hi-hat elements extracted from dense drum loops and make them entirely your own. These don't have to form the sole basis of your own rhythm, but can instead sit behind other programmed hits to add interest.

In the video, we begin with a full drum loop containing a kick, snare and hi-hats. We slice up the audio file to isolate only the hats, then edit and process them to help sit this groove behind our own custom drum hit samples.



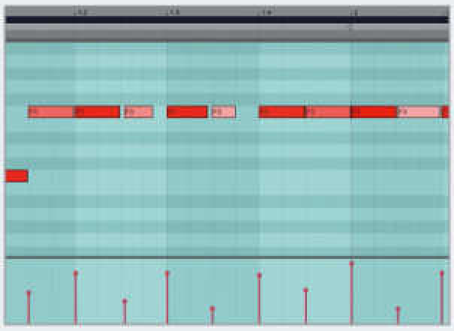
2. Programming hi-hat 'choke'

> A drummer can cut short, or 'choke', an open hi-hat sound using the pedal. In most dedicated drum software, programming a closed hi-hat note shortly after an open one will abruptly cut off the latter's tail. If you have one, it may sound more realistic to cut off the first sound with a 'pedal' sample. In a sampler, all sounds assigned in a 'choke group' will cut each other off, so put your hat sounds in one choke group.

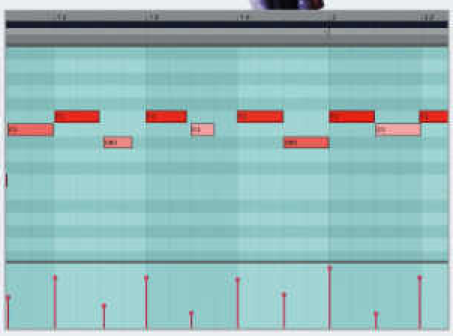
3. Velocity and articulations

> Drum ROMplers do a great job of recreating the sound and tone of a realistic drum kit, but you need to put in a little effort to replicate the performance and timing imperfections of a human drummer. Here's how to do it...

1 > Here's a basic live drum beat we've programmed using Toontrack's EZdrummer 2. The MIDI notes of our hi-hat pattern feature no velocity or timing variation at all, giving a robotic feel. A real drummer would be unable to play with this degree of inhuman accuracy.



2 > We program in velocity variations through the pattern. Accented notes have higher velocities, while notes between have lower ones, mimicking how a live drummer might play. We also move some notes slightly 'off the grid' to humanise our hi-hats.



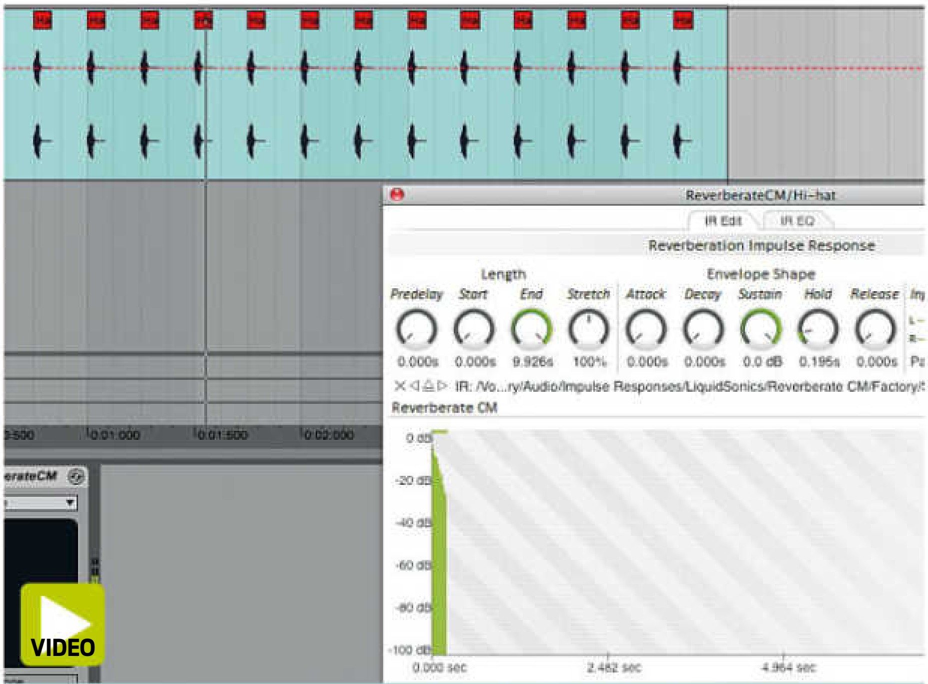
3 > Some drum instruments offer hi-hat articulations to emulate a drummer hitting the hats in different ways, such as with the tip of the stick or the 'shank'. We've added this variety to our groove by alternating MIDI notes, each one triggering a different articulation.

4. Brightening hi-hats' transients

> SKnote's Snap is a unique and handy transient enhancer tool. It's part of the **cm** Plugins collection available with this bookazine and every issue of *Computer Music* magazine. In the video, we begin with a dull hi-hat sample, possibly sampled from an old record. It has character but lacks the sharp sizzle that more modern sounds contain. To remedy this, we load Snap as an insert on the hi-hat channel and reset both Hit and Body dials to zero.

A very slight positive Body amount applies a high-shelf boost to the hi-hat. Now, by raising the Hit parameter, we increase the amount of transient affected by this high-shelf boost, adding brightness only to the hat's initial attack and leaving the rest of the tone unaffected. This keeps the sound's initial tone and character while adding some much-needed 'snap' – a useful effect that's difficult to achieve with regular EQ or transient shaping.

Be careful not to overdo this plugin's effect, as often you'll only need to apply subtle settings for slight transient brightening of your hats.



5. Adding depth with subtle reverb

> We're used to hearing real-world sounds reflecting off nearby surfaces, creating tight echoes – something that dry hi-hat samples can lack. To remove 'dryness' and place hats in the mix, add a tiny touch of short room reverb. It should be a very subtle effect, but it goes a long way in adding a professional sheen to electronic hi-hats.

6. Super-wide hi-hats

> To add width to sounds, many producers will employ the Haas effect. To do this, take a signal, duplicate it, then pan one copy hard left and the other hard right. Now, when one side is delayed by only a few milliseconds, our ears are fooled into hearing one single, super-wide sound source.

We can also use this trick with two similar – but not identical – hi-hat samples. Simply choose two that sound roughly the same, then pan one hard left and the other hard right. No delay is needed, and phase cancellation is minimised because the two hats are different. If this effect is too extreme for you, remember that careful stereo placement of hi-hats through subtle panning can inject real width and space into the high end of a mix. Listen to the video using headphones or quality monitors to hear these effects most clearly.

7. Removing treble harshness

> A brittle, tinny top end is a sign of an amateur mix. Try removing treble harshness from your hi-hat elements like this…

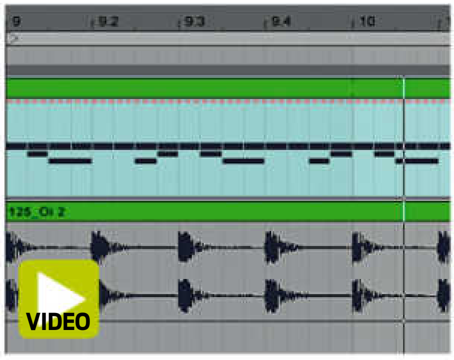
1 > Just as excessive low frequencies in a mix can clog up and muddy the low end, treble and presence areas can often clash and build up, which can cause fatiguing brightness when several sounds compete for the same space in the mix.

2 > The 2-6kHz upper-mid region can add bite to cymbals, but beware of excessive presence above this point – overdo this area, and you could make your mix sound thin. Here we're using an analogue-style EQ to broadly boost the 5kHz area of a hi-hat, while shelving down the 10kHz region.

3 > Saturation can add drive and remove shiny treble frequencies, adding more grit and bite to hi-hats. A wet/dry mix control is particularly useful in this scenario, allowing you to maintain some of the unprocessed signal's brightness while blending in upper-mid crunch and thickness.

8. Applying swing

> Most DAWs now offer some kind of swing function for a jazzier feel, moving the notes found on every second and fourth beat slightly later – listen to the 'lazy' effect our swing setting is giving to our 16th-note hi-hats in the video. Some producers swear by swing grooves captured from old-school hardware such as Akai's famous MPC series and the E-mu SP-1200. Whether the swing functions of these machines are actually 'special' is another issue, but if you'd like to try them for yourself, Live has a built-in selection of old-school drum machine grooves in the Core Library's **Swing and Groove** folder.



9. Creative sidechaining

> In the video, we've made a rigid and unnatural eighth-note hi-hat pattern. We could dive in and alter volumes and lengths in the piano roll or in a sampler, but there's an even speedier solution using sidechain compression

A compressor is placed over the hi-hat channel, and we set up the tom loop as a sidechain trigger. The tom loop is triggering the compressor's gain reduction over the hi-hat each time it plays, ducking the hi-hat channel down and providing movement. We can mute the tom channel and still hear the dynamic variation in the hats.

10. Real electronic hats

> Electronic drums often contain the weight and smack required for a punchy modern mix, but they can sound rather static and lifeless in comparison to a live drummer's performance. Realistic hi-hat layers can therefore add some subtle energy and realism behind electronic beats, bringing them to life.

In the video, we begin with a dry DnB drum loop. The addition of a simple hi-hat groove from EZdrummer 2 adds real character behind our sample-based beat. Note that our drum ROMpler adds in subtle variations like a real drummer, so if we want our hats to play back the same each time, we can simply render out a loop of these hats and re-import the audio for consistent playback. cm

SMOOTH OPERATOR

Ableton's FM-powered synth is bursting with sonic possibilities far beyond typical DX7-alike tones. Fire up Live, load up Operator and we'll show you how!

> **Frequency modulation (FM) synthesis allows for the creation of a huge array of sounds, ranging from tuneful and harmonic to dissonant and chaotic. This is all made possible through the interplay between so-called oscillators designated as carriers and modulators. The frequency of one such oscillator – also called an 'operator' in FM terminology – is modulated using another that is also in the audible range, resulting in a more complex waveform. Even using simple sine waves as the raw waveforms of the operators can result in timbres that are rich in harmonics. Increasing the amplitudes of the modulators will give increasingly harmonic-rich signals.**

Taking the template laid down by classic digital FM synths like Yamaha's DX7 and SY77 considerably further, Ableton adapted this concept for their own Operator instrument, which is a $99 add-on for Live, or included as standard with Live Suite. Combining the concept of FM with both additive and subtractive synthesis (which, curiously, do not cancel each other out), Operator is your gateway to a universe of complex timbres, all dialled in via an interface that is intuitive and simple to, well, operate.

With a solid selection of basic waveform types to choose from and the option to create your own using the waveform editor, you can arrange Operator's four oscillators in several predefined routings – known as 'algorithms', in a nod to the very similar system employed on the DX7 – to synthesise a wide variety of musical and non-musical tones.

In this guide, we will explore the breadth of sonic possibilities of Operator, using different algorithms for different use-cases. From designing kick drums to sculpting evolving pads and expressive lead sounds, each of the tutorials will introduce you to further features of Operator as well as giving you the techniques to use them for designing your own sounds from scratch.

DOWNLOAD
Get the tutorial files to follow along on your PC/Mac at **vault.computermusic.co.uk**

Operator's ancestors: FM history

Frequency modulation has a history with music technology owing to its use since the 1930s in broadcasting FM radio. However, John Chowning takes the credit for inventing FM synthesis. Whereas FM radio uses super-high, inaudible carrier frequencies in order to transmit content (music, speech, etc) that must be 'decoded' upon reception into a signal in the audio range, FM synthesis operates very much in the audible range, as you'd expect.

Chowning came across the concept during the 60s while studying the characteristics of vibrato at Stanford University. Vibrato is frequency modulation: we use one oscillator – an LFO in the inaudible range – to modulate the pitch of an audible oscillator, producing a noticeable 'wobble' in pitch. Chowning noticed that increasing the rate of the LFO into the audible range produced a complex new tone, and further pursuit of this phenomena led him

> "Chowning came across the concept during the 60s"

to formalise – and patent – its use for sound creation as FM synthesis. In 1967, he became the first person to compose a complete piece of music using FM for all sound generation tasks. Check out **bit.ly/1aD91pc** to hear him explain the discovery in his own words.

Famously, Yamaha licensed Chowning's invention and enlisted his assistance in creating a series of instruments based upon his technology, eventually leading to the legendary Yamaha DX7, which is one of the best-selling synths of all time. It was a very difficult device to program, though that didn't deter determined synthesists from mastering it, using its powerful operator-based synthesis and multiple routing 'algorithms' to sculpt spectacular new sounds.

As we just alluded to, the oscillators in the DX7 are referred to as "operators". Inspired by old hardware FM synthesisers, and adapting these concepts to the modern DAW environment, Ableton came up with their own hybrid approach to synthesis, combining FM with additive and subtractive synthesis, introducing Operator in 2005. While there have been major improvements to the device over the years – such as the major makeover in 2008 seeing drawable wavetable features added alongside new filter types and routing options – the interface has remained consistent throughout the years, following Ableton's philosophy of fusing depth and usability into one creative whole.

> Step by step

1. Designing a kick drum in Operator

1 > Create a new **MIDI track** in Live. Load an **Operator** device from Live's browser. Create a one-bar MIDI clip and put a **C3** on every quarter-note. Play the clip and you'll hear a simple sine wave on every beat – this dull sound is the initialised patch, the starting point for our final kick drum. First, we need to change the way that Operator routes signals.

2 > Clicking the bottom-right section (including Time, Tone and four coloured squares) opens the **Global** control section. At the top of this section, choose the rightmost diagram (four horizontal squares) to change Operator's routing to **Additive**. This gives us four independent oscillators (with no frequency modulation) that we can layer to create a richer Kick sound.

3 > To start shaping the sound, first set all the oscillators to **Fixed** in the left section. The frequency of each will now be the same, regardless of any incoming MIDI note. Set the Oscillator A's **Frequency** to **50Hz**, then back in the centre panel, set its **Attack** to **12 ms**, **Decay** to **300 ms** and **Sustain** to the lowest value (**-inf dB**). **Kick1.wav** showcases this foundation of our Kick.

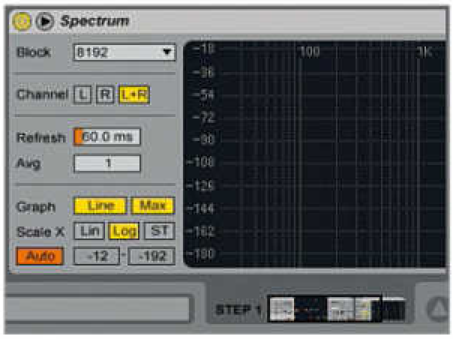
4 > We'll use Oscillator B to add more body and a beater (click) sound to the kick. Set its **Frequency** to **94 Hz** and bring the **Level** up to **0.0 dB**. Leave **Attack** at **0.0 ms**, set **Decay** to **100 ms** and **Sustain** to **-inf dB**. Then set **Phase** to **5%**. Shifting the phase off the zero-crossing point like this is what will cause the clicking sound.

5 > We'll add more body to our Kick with a third oscillator. Change Oscillator C's waveform to **Sw3** (a saw wave). Adjust the envelope settings – set the **Attack** to **25 ms**, **Decay** to **600 ms** and **Sustain** to **-inf dB**. Then, set **Freq** to **42 Hz** and bring the **Level** up to **-16 dB**. As you can hear in **Kick2.wav**, this adds more low end to the sound.

6 > Finally, to add more punch and character to the kick, turn on the pitch envelope and set **Pitch Env** parameter to **15%**. To prevent the pitch envelope from modulating Oscillator C – the sound's body – remove Oscillator C from the Dest. A section of the pitch envelope as above. Hear the finished result in **KickFinal.wav**.

> Step by step

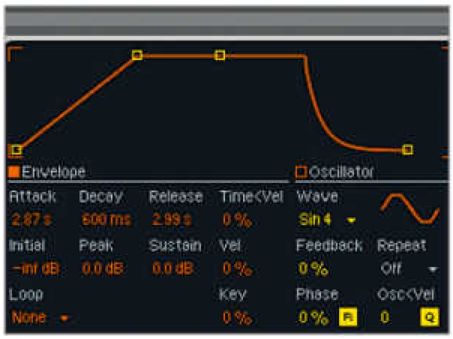
2. Analogue-sounding bass with Operator



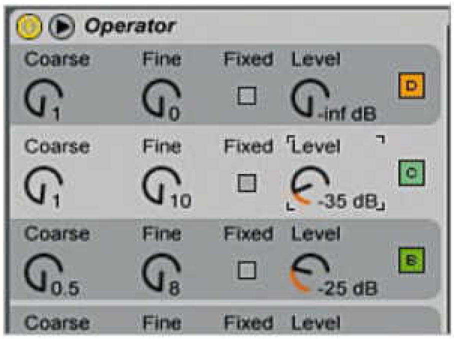
1 > Drop an **Operator** device onto a new MIDI track. We're going to use Frequency Modulation techniques to create a bass sound, so it's helpful to visualise the changes we make to the sound over time. Drag a **Spectrum** audio effect device after Operator in the chain.

2 > Click on the bottom-right section to open the global controls. We'll use a signal path in which Oscillator A is our carrier signal (ie, the audible oscillator), modulated by oscillators B, C and D. Choose the 'three on one' schematic. Also, since we are creating a monophonic bass sound, change the number of **Voices** to **1** and set the (bottom-right) **Volume** to **0dB**.

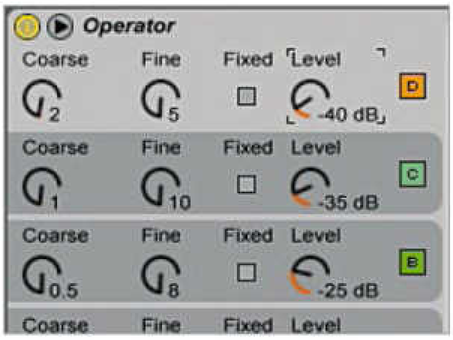
3 > For Oscillator A, we select the default **Sine** wave – this will mostly be used for our sub-bass information. Change **Coarse** to **0.5**. This will tune the oscillator to the lowest octave possible. Set **Attack** to **0.35 ms**, **Decay** to **590 ms**, **Sustain** to **-6.0 dB** and **Release** to **3.00 s**. Hear the sound start to take shape in **Bass1.wav**.



4 > We employ Oscillator B to add more harmonics to our timbre. Change the waveform to **Sine 4** – a retro, 4-bit, C64-style sine wave. Set its **Coarse** to **0.5**, and bring up the **Level** to **-25 dB**. Set **Fine** to **8** to fatten up the sound by detuning the modulator. Set **Attack** to **2.87 ms** and **Release** to **3.00 ms**.

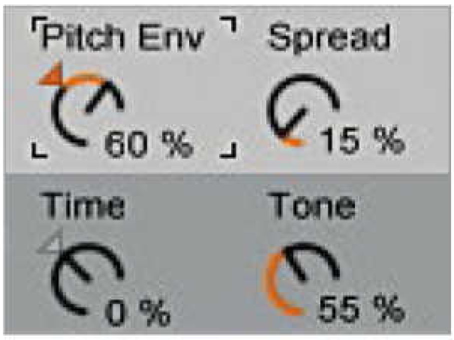


5 > For our second modulator, Oscillator C, we will use a harmonically richer waveform, **Saw D** (D stands for Digital), which is particularly good for bass design. Set **Fine** to **10** and bring the **Level** up to **-35 dB**. You can hear that this detuning causes more movement. Set **Attack** to **200 ms** and **Release** to **3.00 s**.

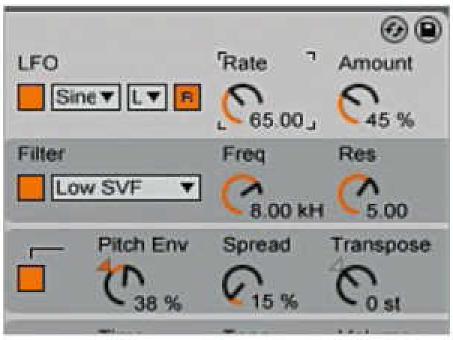


6 > We use Oscillator D to add some excitement to the timbre during longer sustaining notes. Choose another **Saw D** waveform and set a long **Attack** and **Release** time of **10.0 s**. Set its **Coarse** to **2**, **Fine** to **5** and bring the **Level** up to **-40 dB**. You should be able to hear subtle modulation with longer notes.

7 > Next, we'll use the pitch envelope generator to imbue the sound with more punch and attack. Change the **pitch envelope** parameter to **60%** and stop the envelope from affecting oscillators B, C and D in the Dest. A section. Now the pitch envelope will only be applied to the carrier signal (oscillator A). Set **Peak** to **+30 st** and **Decay** to **200 ms**.



8 > We add some stereo width by setting the **Spread** to **15%**, panning the oscillators a little. One of the issues with FM synthesis is that you can easily end up with a lot of high harmonics. Operator's **Tone** parameter lets you control the amount of these easily. In **Bass4.wav**, we've brought it down to **55%** just to control the top-end.



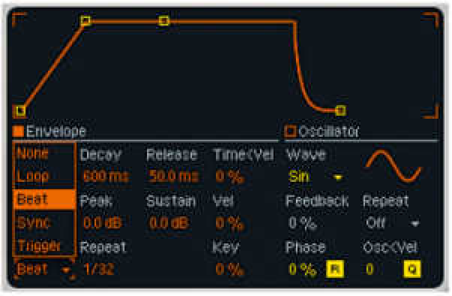
9 > Finally, to add some filter modulation, Turn on the **LFO**, then set its **Rate** to **65** and **Amount** to **45%**. Set a long **Attack** and **Release** of **9.00s**. De-assign the LFO from all the Oscillators and assign it to **Filter** in the Dest. A section. Set the Filter's **Frequency** to **8.00 kH** and **Res** to **5**. Hear the finished product in **BassFinal.wav**.

Operator's Envelope Loop Mode

Envelope generators are one of the key tools for creating spectacular and interesting sounds with FM and additive synthesis, and Operator has seven of them – one for each oscillator, and dedicated envelopes for filter, pitch and LFO – giving you a wide range of sonic vistas to explore.

One of the most useful features of Operator's envelopes is their Loop Mode. When an envelope is in Loop Mode, it'll retrigger upon reaching the sustain stage as long as the note is being held. The loop can be set to run at a free rate or synced to the project tempo.

The Loop Mode has four modes to choose from. The first is Loop, where you can set the envelope's Time parameter in milliseconds – after reaching the end of the decay stage, the synth will then wait for the specified time before looping back to the beginning of the attack. Since this value will be affected by the synth's global Time parameter (in the lower-right of the interface), you can achieve interesting results by modulating the global Time value using the LFO.



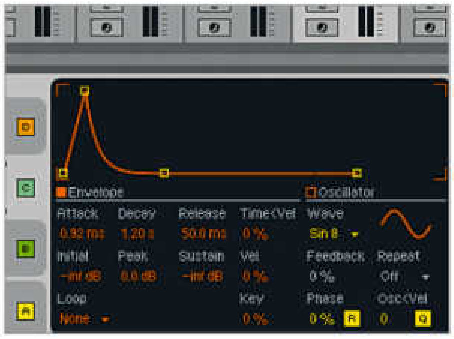
Operator's seven envelopes don't just have to be one-shots – loop them in one of four modes

The next modes are Beat and Sync, which reset the loop using rhythmic values such as quarter-notes or eighth-notes, synced to the tempo of your project – particularly good for the creation of rhythmic patches. In Beat mode, while the loop length itself is perfectly quantised, if you play the note slightly out of time, each repetition of the loop will be off-grid by the same amount. In contrast, Sync mode snaps the loop repetitions to the nearest 16th-note for a tighter, more musical sound. Note that Sync mode only works if the set is playing, otherwise Sync and Beat modes will behave the same way.

Finally, Trigger mode is ideal for percussive sounds, triggering the envelope while ignoring a note-off message, meaning that the length of the sound is unaffected by the length of time for which the note is held down.

> Step by step

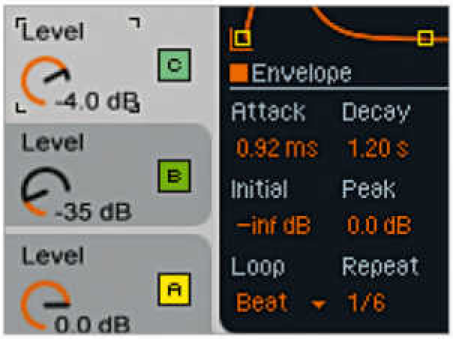
3. Creating ambient pads using Operator

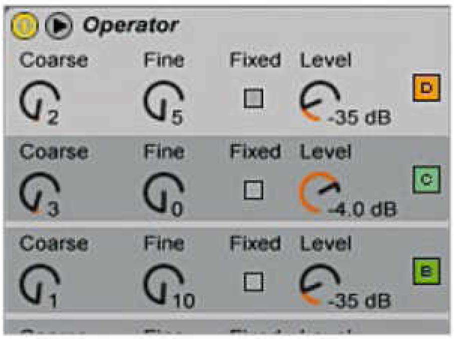


1 > Drag an Operator device onto a new MIDI track in Live. Click on the global section on the bottom-right. For this sound, we'll use an algorithm that has two carriers and two modulators. This means Oscillator A and Oscillator C are going to be our audible signals, and they'll be modulated by Oscillator B and Oscillator D respectively.

2 > First, let's create a sustained pad sound. Change Oscillator A's waveform to **Sq4**, and adjust its **Attack** to **6.00 s** and **Release** to **10.0 s**. Now to add some unison using Oscillator B. Bring its **Level** up to **-35 dB** and detune it by setting **Fine** to **10**. You'll hear the detuning cause some movement, especially when playing chords as in **Pad1.wav**.

3 > Now lets add our second sound. Set Oscillator C's waveform to **Sin 8** – a retro sine wave that has more character than a more 'hi-res' sine wave. Set its **Coarse** to **3**, **Attack** to **0.92 ms**, **Decay** to **1.20 s** and **Sustain** to **-inf dB**. Bring **Level** up to **-4.0 dB**.





4 > One of the most useful features of Operator is its ability to loop envelopes. Select Oscillator C's tab, click **Loop**, select **Beat**, and set the **Repeat** parameter to **1/6**. In **Pad2.wav**, you'll notice that the envelope re-triggers itself while the note is being held. This effect is particularly interesting when you play multiple delayed notes.

5 > Now to add some harmonics to our second sound for more character. Change Oscillator D's waveform to **Tri**angle, set its **Coarse** to **2** and **Fine** to **5**, and bring its **Level** up to **-35 dB**. This will add some higher harmonics to our bell sound – helpful in giving more brightness to the sound in higher octaves.

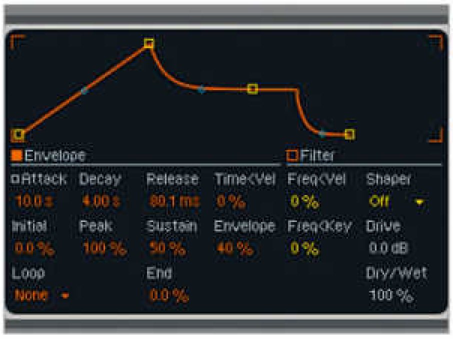
6 > Some final touches. Set **Spread** to **40%** for stereo width, and bring the master **Volume** to **-17 dB** to prevent clipping. Drop a Reverb Effect after Operator, setting its **Decay Time** to **6.00 s** and **Dry/Wet** to **30%**. The sound's effect is best brought out by slowly introducing sustained notes from different octaves. Check out ours in **PadFinal.wav**.

> Step by step 4. Phat sawtooth chords with Operator

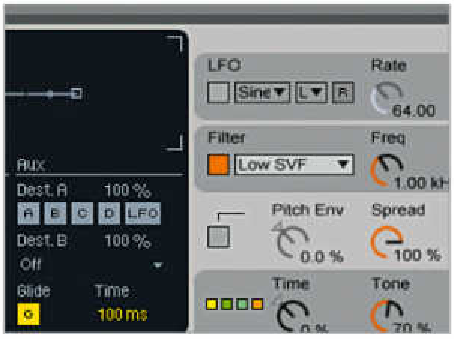
1 > To create a lead chord sound, we open a new instance of Operator using the additive algorithm. This mean we'll have four independent (shown horizontally) audible oscillators without frequency modulation. Click the global section at the bottom right, select this routing and set **Voices** to **1** for a monophonic sound.

2 > Choose **Sw32** – a sawtooth wave with 32 harmonics – for every Oscillator. Bring the Level of each oscillator up to **0.0 dB**. Also, set the global **Transpose** value to **-12 st**. This will transpose the whole patch down one octave.

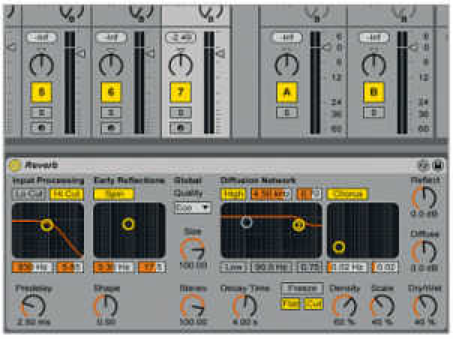
3 > We'll use the **Fine** control of each oscillator to generate a chord, but tune them slightly off to achieve a unison effect. Set Oscillator B's Fine to **198 cents** instead of 200 (a major second), Oscillator C's to **500 cents** (a fourth) and Oscillator D's to **790 cents** (a minor sixth).



4 > Now lets use the filter envelope. In the filter section, set **Freq** to **1.00 kH** and **Res** to **3.00**. Set the Filter Envelope's **Attack** to **10.0 s**, **Decay** to **4.00 s** and **Sustain** to **30%**. To apply the envelope to the filter, set the **Envelope** parameter to **40%**. This will open the filter slowly, resulting in an evolving sound, as heard in **Lead1.wav**.



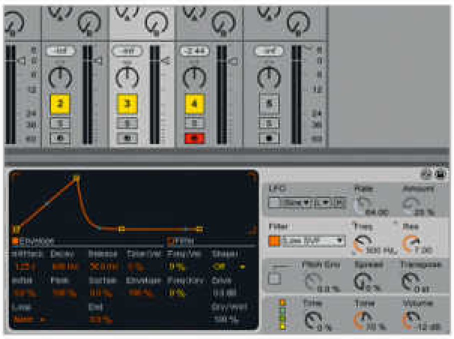
5 > Click on the Pitch section. Turn on Glide by clicking the yellow **G** button, and set its **Time** parameter to **100 ms**. This will slide the pitch between notes that overlap. Set **Spread** to **100%** – this creates a richer stereo sound by detuning the left and right signal.



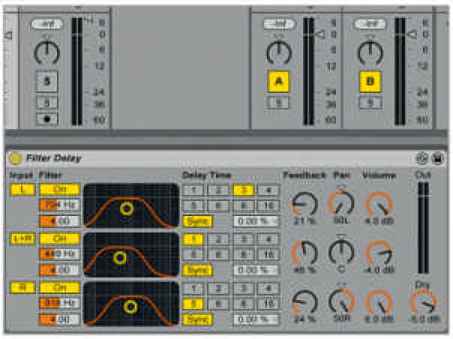
6 > The lead patch is ready, but some effects processing can help achieve a more interesting result. Drop a Reverb effect after Operator. Set its **Decay Time** to **4.00 s** and **Dry/Wet** to **40%**. The result is a smoother sound, especially after the release stage, now that we've added a longer tail. **LeadFinal.wav**.

> Step by step 5. Rhythmic FX design in Operator

1 > Drop an Operator device onto a new MIDI track. Change Oscillator A's waveform to Noise White (**NoW**). Set its **Attack** to **0.40 ms**, **Decay** to **755 ms** and **Sustain** to **-inf dB**. Set envelope **Loop** to **Beat** and the **Repeat** parameter to **1/16**. This will re-trigger the envelope generator on every 16th-note.



2 > Set the Filter's cutoff **Freq**uency to **500 Hz** and **Res**onance to **7.00**. Now set **Attack** to **1.25 s** and apply the envelope to the filter by setting the **Envelope** parameter to **100%**. This opens the filter gradually, while the high resonance value creates an exaggerated sweeping effect.

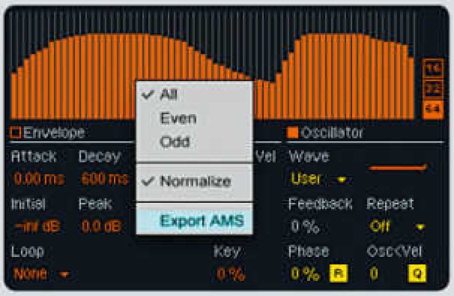


3 > Live's Filter Delay can be used to introduce rhythmic movement to the sound. Drag one in from Live's browser and drop it after Operator. Operator's noise generator is also a really good tool for creating percussive sounds such as claps and hi-hats. Check out the end result in **SFX.wav**.

Tips and tricks

CUSTOM WAVEFORMS AND SAMPLER

In addition to the simple waveforms available in Operator, you can create custom waveforms using the waveform editor. You can also save these custom waveform for later use as .ams files by right-clicking the waveform editor and choosing Export AMS. One of the advantages of this feature is that you can drag these AMS files into other Ableton devices, such as Sampler, for use with their modulation capabilities and other processing functions.



Export your custom operator waveforms as AMS files for later use or to import into other Live devices

AUTOMATE THE ALGORITHM

Usefully, Operator lets you map the algorithm selector to a Macro control or a MIDI controller. As you already know, changing the algorithm has a significant impact on the sound, since it changes the global behaviour of the instrument, so there's plenty of mileage to be had experimenting with changing the algorithm on the fly. Try mapping the algorithm to an LFO for potentially far-out results.

SAMPLE AND HOLD

Operator's LFO can be set to output a range of waveforms, and one of the more interesting of them is Sample And Hold (S&H). This signal uses random values chosen at a rate determined by the LFO, to create sci-fi-style sound effects and introduce random, unpredictable modulation to the sound.

THE FIFTH OSCILLATOR

Typically, Low Frequency Oscillators (LFO) have subsonic frequency values. This means we will only hear their effect and not their actual sound. By setting Operator's LFO range to HIGH, you can push it into the audible range – as far as 12kHz. You can think of Operator's LFO as a fifth oscillator.

Take the chisel to your waves with Operator's waveshaper

BUILT-IN WAVESHAPER

Operator's Filter panel features a built-in waveshaper. Select the waveshaping curve via the Shaper menu to access the Drive and Dry/Wet controls.

USING THE COARSE PARAMETER

One of the main controls in Operator's oscillators is the Coarse parameter. Note that Coarse is a frequency multiplier and not an octave selector. This means that the frequency of the oscillator will be multiplied by its Coarse parameter. So if Coarse is at 1, an A4 note will be 440Hz; if Coarse is at 2, the frequency will be 880Hz (an octave higher: A5); if Coarse is set to 3, the frequency is 1320Hz – E5; and so on.

Is the top end of your FM patch grating on you? Cut out high-frequency artifacts with the Tone control

ANTI-ALIASING

Digital FM synthesis techniques can result in aliasing artifacts when creating timbres with a lot of high frequency information – sometimes desirable, sometimes not, depending on the patch you're creating. To this end, Operator has a togglable high-quality antialiasing filter in the Global section, as well as the global Tone filter, which can be used to curtail runaway treble frequencies as much or as little as you like.

GLOBAL TIME CONTROL

One of the other useful global controls in Operator is the Time parameter. Using this knob, you can scale all the envelopes' timing up and down. This can be used to change the sound drastically, or modulated rhythmically using the LFO for a more dynamic effect.

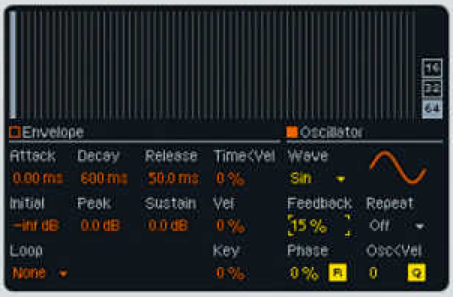
GO NEGATIVE

Playing with Operator's envelopes, you'll notice that some parameters can be set to negative values; for example, you can apply -50% in the Filter Envelope. Think of this as flipping the envelope upside down, so that a slow attack causes the filter to gradually fall from its initial position, rather than rising, as with a normal, positive modulation setting.

Get experimental and mess with your envelopes even further by applying negative amounts

CPU SAVING TIPS

Operator is an optimised native Ableton instrument, and its CPU usage is pretty low. Nonetheless, when using several instances of it at once in a project, you might find that you need to cut down on their CPU usage for better overall performance – particularly when performing live. You can do this by simply disabling any of Operator's features that you're not actually using, such as the Filter or Spread, for example.



Push an oscillator into self-modulation by adjusting the Feedback parameter in the central section

FEEDBACK

We already know that we can modulate an oscillator's pitch using one or more of the other oscillators, but there is another option: any oscillator that's not being affected by the others can modulate itself. You can control the amount of this modulation by adjusting the Feedback parameter in the envelope section of each oscillator. cm

Operator's LFO can be pushed into audible frequencies – useful when four oscillators aren't enough

MASTER BUS FX

Make better breakdowns, DJ-style effects and more with our essential guide to master mix processing

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> We all use audio editing and plugins on individual sounds and tracks to give them creative flair, but how many of us think to use such processes on the main output or for processing our final masters? This could involve anything from hands-on edits to simply adjusting the volume curve in order to drop the overall level a little at certain points, but by processing the full track rather than select parts of it, you get very different results. In a similar way to how a compressor placed across a bus creates greater cohesion than the same compression applied separately to each part within the group, master bus processing can give a track a greater sense of cohesion, impact and power.

Of course, as with any music production process, there are a number of options and considerations to take into account. For example, there's no point spending all your time carefully crafting a perfect mixdown only to destroy the carefully tuned balance by over-processing it. Things like the order in which the effects are placed become even more important when dealing with your master output, too, as the wrong signal chain can cause things to become terribly thin or hopelessly muddy very quickly. But as long as we keep these basic considerations in mind, there aren't really any other limits or restrictions.

Over the next few pages, we're going to go through some useful master bus processing techniques. Almost all of them will work with even the most basic plugins, but don't hesitate to try them with more interesting effects, like our own Vengeance Philta CM and Artillery 2 CM. Producers have been using some of these tricks since the days of tape recording, so there's really no excuse not to in today's virtual studio. And you don't have to just use these effects with your own finished tracks – whether doing a DJ mix in Live or simply creating custom edits of tracks to play out, all of these techniques can be applied just as easily to other people's music as they can your own.

Let's start by looking at one of the most basic but useful types of master bus processing: creating a build-up at the end of a breakdown. The obvious approach for many is to use sounds that can be easily repeated quickly, such as a synth hit or snare drum. Perhaps a building delay will be applied to such a sound, with the delay level and feedback rising to create extra intensity as it progresses, possibly complemented by a building reverb, too. Then there's the classic rising pad, whereby white noise and/or a synth-pad are made to 'rise up' through the opening of a low-pass filter.

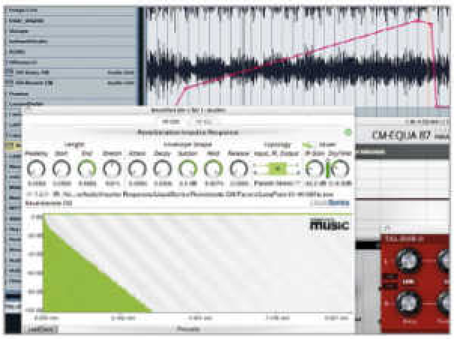
These are all still wholly valid techniques (indeed, the last one works brilliantly in tandem with the following), but for the best results possible, we approach our builds via the master output bus. Using just a delay, an EQ/filter and a reverb, we can create huge breakdowns in a matter of minutes. And, keeping things simple, we limit ourselves to just a few controls, so we can manually record the movements of each plugin, making things much more expressive.

The whole approach is very similar to the way DJs manipulate effects to enhance track build-ups, and of course it can be done in exactly the same way, but we think that for everyday production, this combined approach – taking control of single tracks in addition to creating the overall build at the master output – delivers the best results. That said, whenever you mess around with your master output you have to be careful not to compromise the sound quality, particularly avoiding distortion and overdrive limiting, which no amount of mastering will be able to undo. Here's how to do it right…

The master bus filter sweep played a big part in defining the sound of Norman Cook's output as Fatboy Slim

> Step by step Creating a big build-up

1 > Import the audio file **1 Original unprocessed buildup.wav** into a new project, then insert the free TAL-Dub II plugin (**kunz.corrupt.ch/products/tal-dub**) into the master output. Try a synced eighth-note delay. Next, add eaReckon CM-EQUA87 and engage the low-cut filter using automation at the start of the breakdown, then disengage it a fraction before the end of the break.



2 > Now add LiquidSonics Reverberate CM and automate the **Dry/Wet** control so that it's dry until the start of the breakdown, then quickly raised to about **5-8%** for the duration of the break, ultimately rising up to about **30%** over the last four bars of the break.

3 > Raise the delay mix slowly from totally dry to about **5%** wet up to the last eight bars of the break, then draw in a jump up to about **40%** wet up to end of the break. You can either set a fixed **Feedback** amount or have that rise up at the end, too.

4 > Tweak the automation lines to give them an increasing curve, getting higher and faster towards the end, so that the energy rises quickly to a final peak. The trick is to keep things quite clear right up until the crescendo.

5 > Things are getting pretty heavy at the end of the breakdown, so now we sweep the low-cut filter cutoff frequency up over the last four or two bars. That way, we feel the intensity but still have somewhere for the track to go when it kicks back in.

POWER TIP

>Limited company

It should be clear by now that the level can rise considerably when doing this sort of thing, so it's important to have a compressor or limiter on Live's main output after these building effects, which we'll refer to from hereon as a "safety" compressor/limiter (as opposed to the one we use during mastering, to make our track sound cohesive and phat). It's also worth placing a basic gain control and meter before them, ensuring that the effects aren't overloaded internally, thereby adding distortion.

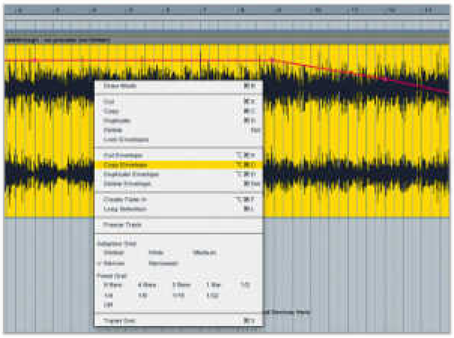
> Step by step Creating a big build-up (continued)

6 > Time now to carefully tweak the end parts. Our automation curves have been building up and up throughout, but over the last bar we want the curves to ramp back down to normal, to make the track sound like it's 'sucking' back in. The curves depend on the particular track, so trust your ears – just make sure the first beat after the break hits cleanly.

7 > It's not an essential step, but sometimes we like to lower the threshold on the master bus safety compressor right at the peak, then quickly return it before the first beat. This intensifies the effect and emphasises the greater dynamics when everything kicks back in.

8 > Have a listen to your work so far. It shouldn't build too much or too soon, and it's important to tweak the various effects' parameters so that they compliment each other. See *The right order*, below, for some tips on the relationships between the plugins.

9 > Let's work some saturation into our breakdown. Add the Sonimus Satson plugin before the compressor, raise the **Gain** a little at the start of the break using automation, then gradually ramp it up as things begin to get more intense. The result should be a bit more atmosphere when things drop.



10 > Copy the automation to other drops in the track, but with less extreme settings, particularly where they're shorter. We also bypass all of these effects until just before the start and end of the breaks to ensure our master output isn't compromised throughout the rest of the track. (Audio: **Build walkthrough – with process.wav**)

POWER TIP

>Mastering in mind

Generally speaking, we like to place the master bus effects described in these tutorials before the conventional mastering chain, as it pulls everything together better, but there's nothing to stop you placing them afterwards – just ensure there is some kind of protective limiter placed at the very end to prevent clipping. Placed after the mastering chain, master bus processing is more like a DJ playing with your track rather than the effects being a part of your track, so if that's the sound you're after, why not give it a go?

The right order

The order in which we place our master bus build-up plugins has a huge impact on the sound. While you shouldn't just automatically use the same order for every track (it's always worth mixing things up), there's nothing wrong with starting at a logically laid out default setup – the delay-filter-reverb configuration we've used in the above walkthrough is a perfect example.

Why do we call this 'logical'? Well, we place the reverb last in the chain for a variety of reasons, all to do with the way it interacts with the other effects. For example, if we place the filter after the reverb, the reverb tail is filtered too, creating a much sharper, more unnatural effect. Of course, sometimes we might want this, but normally, the soft smoothing effect of the reverb tail continuing slightly after the filter sweep has cut off the sound is preferable and more atmospheric.

Similarly, placing the delay after the reverb can create a slightly messy, unnatural sound as the reverb tails are also delayed. But again, this might work for your breakdown.

We place the filter after the delay, as when the delay starts feeding back on itself, things can begin to get messy in the bottom end. Thus, we want the delay tails to be filtered too, especially towards the end of the breakdown, where we want very accurate control of the bottom-end weight of the track right before everything kicks back in, in order to maximise impact.

Delay > Filter > Reverb

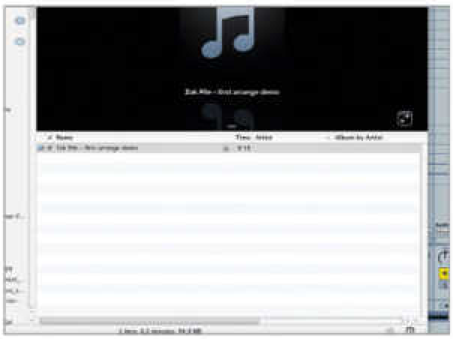
Doubling up

Some of the best 'master edit' effects were devised by hip-hop DJs and turntablists. For example, live flanging was generated by running two copies of the same track at the same time (the natural turntable speed drift creating the flanging). They would also generate 'doubles' during live play by manually scratching and dropping in a second copy of a track from a point just an eighth note in front of the one currently playing, then pulling the crossfader back and forth to create the effect. This is quite an extreme effect that can have great impact in certain tracks, but for many people, it's a surprisingly confusing one to recreate. Obviously, one way is to slice every region in your arrangement at the point you want to use the effect, so that you have a discrete chunk of arrangement an eighth-note long at the start of, say, every eight bars, then slice and delete the preceding eighth-note value and copy the eight-note chunk back. This is quite fiddly to do, though, as you can easily cause all manner of zero-crossing clicking (on any of the parts, of which you might easily have 20 or more playing). It can also end up sounding too heavy, as this trick is designed to work in conjunction with the crossfader and works best when the first hits are a little quieter.

In fact, it's much easier to do it using an audio file of your finished track. Ableton Live is particularly effective for this sort of editing, as it's to quick and easy to navigate. It even features an assignable crossfader, but as long as you copy the doubled edits to a new channel and simply lower the level on that track a decibel or two, you can get away without actually creating the crossfade in most cases. OK, let's do it…

Turntablists certainly know a thing or two about 'master edits', so study their work and exploit their tools

> Step by step DJ-style doubling

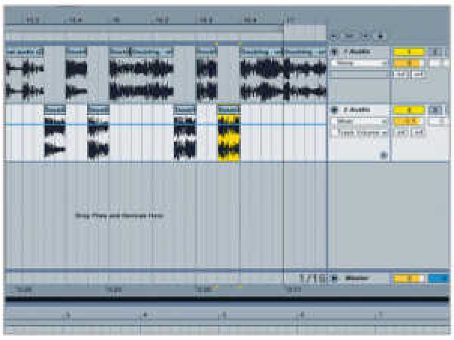


1 > We start by bouncing our project down to a single audio file (**1 Doubling walkthrough – no effect.wav**), listening to the track and deciding on the points at which we want to add our doubles. We don't want to overdo it – this isn't a DJ set, after all. Just one at the start of a middle eight or each verse should be enough.

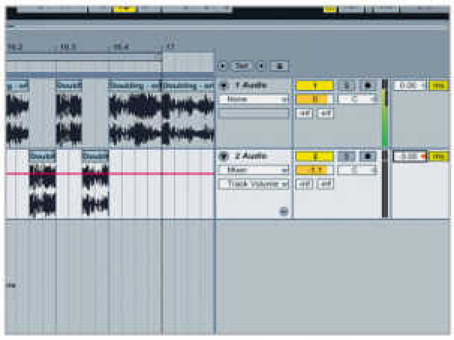
2 > We put our mix on an audio track in a new project, being sure to set the tempo correctly and line the start of the first beat up with a whole bar value in the arrange page, so that we can use the timeline for meaningful reference.

3 > Now we go through and make the slices. We need to chop and create a new region from the first eighth-note length of the section to be copied. We might want to create some small overlaps later, but for now it's easier to keep the lengths and cuts precisely to the grid.

4 > Delete the previous eighth-note length section and copy the section to the eighth-note immediately before it on the channel below.



5 > The effect works best when the 'pre'-double is slightly quieter, so reduce the channel level by 1-2dB. There's nothing to stop you doing this by reducing the region's gain if you prefer – the channel level's just easier if you're changing a few of them together.



6 > Live automatically adds short fades to region endpoints, so there shouldn't be any clicks at the crossover points. We pull the pre-double channel back 3ms to introduce a touch of DJ-style looseness. (Audio: **6 Doubling walkthrough – with effect.wav**)

TIPS & TRICKS

DELAY SWEEPS

Insert a delay with a single wet/dry control and create cool edits by sweeping the control at the end of every eight bars. This works best with a fast delay time (16th-notes, for example) and, ideally, a ping-pong delay, which is less obtrusive.

MULTIBAND SIDECHAINING

Try using a multiband sidechain plugin on the master output to create subtle low-frequency pumping. Vengeance Sound's Multiband Sidechain is designed for that very purpose, but Live's own Multiband Dynamics device can do it, too – just apply the compression with a gentle Threshold and Ratio to everything below around 100-200Hz using a sidechain keying signal from the kick drum.

FEED IT BACKWARDS

Precise reverse edits can be jarring because the timing is a little too precise – they often sound better with the reversed material moved a touch early. Add your finished mix to a new project and create the reverse edits manually on another channel, then move them forward in time by 2-10ms, to let the end of the main track overlap the start of the reversed section by a tiny bit.

HANG THE DJ

People are increasingly used to hearing DJs use specifically DJ-orientated effects to enhance the breakdowns in the tracks they play, so why not use them in your own productions? Use DJ software effects in programs like NI's Traktor, manipulating them on the fly, then edit the results back into your master file.

A low-pass filter is your best choice for entering a breakdown, followed by a high-pass at the height of the action

HIGHS AND LOWS

As a general rule of thumb, use a low-pass filter to smoothly transition into a breakdown, and a high-pass filter at the peak of the breakdown to maximise the impact of the track kicking back in.

GET PHYSICAL

You can't beat the hands-on approach, so map all the main controls of your breakdown plugins – such as the delay and reverb wet/dry, and filter cutoff – to knobs and/or faders on your MIDI controller.

KEEP IT SIMPLE

The more extreme the effect, the quicker people will tire of it, so be careful: when adding doubles, delay sweeps, reverse sections and other edits, the temptation is to throw the kitchen sink at the track, then copy it to the end of every eight bars. Don't do it!

DJ-style effects are designed to give instant hands-on results – perfect for our purposes here, then

SEASON OF THE GLITCH

Glitch edits often sound smoother when bounced and placed on a separate channel. Simply record the edit pass using your glitch plugin, then line up the edits underneath the original track, being careful not to create a volume jump between the two channels.

ALL FOR ONE

It can sometimes be very effective to make a big, inclusive build-up but with just one sound (a fast synth sweep, for example) remaining unaffected. Create two tracks in a new project – one dry with your final mix on it and another with the sound that you're going to leave unprocessed. Alternatively, reroute everything going to your master out to an auxiliary channel, then send that and the 'dry' channel to your main output.

YES WE PAN

Using a panning effect in your build-up can work wonders, and a great trick for getting away with extreme left-right panning without creating an off-balance mix is to use a bass mono-ising plug and set it so that the sound doesn't pan below a certain frequency. Making the mono cutoff frequency rise or fall as the speed of the panning increases is a great way to enhance the build.

DOUBLE TIME

Doubling works best where there's a specific sound, vocal or effect that occurs at the start of an eight-bar section. Another interesting place to try them is on the snare hit that falls on the fourth beat of the first bar of eight.

THE BEAT 2 DROP

A good DJ-syle filter edit to try with either a high- or low-pass filter is to sweep the cutoff to remove the top or bottom end over the course of the last bar of eight, then sweep it back up from the first beat of the next bar to the second beat, so that the second beat hits at full frequency. cm

ESSENTIAL TRANSITIONS

Got a few good musical ideas but finding it tough to get from A to B? Bridge the gaps with these ten essential transitional techniques

DOWNLOAD

Watch every video on your PC/Mac to see how it's done at **vault.computermusic.co.uk**

> Transitions and fills can be defined as passages, riffs, rhythmic sections or sounds that connect contiguous musical sections together. You will have heard many of them – indeed, there was probably at least one in the last song you listened to, whether it was a smooth segue between verse and chorus, or an attention-grabbing fill before a drop.

Placing a crash cymbal on beat 1, muting track elements and inserting crazy drum fills are just three of the most obvious and ubiquitous ways to join disparate sections of a song, but arrangement-bridging methods can come in a vast array of forms, depending on the context in which they're used – especially as new plugins emerge and techniques evolve.

Having said that, there are many tried-and-true methods that, when employed correctly, never fail to connect with the listener – and we're going to show you ten of them right here, right now.

Our examples are by no means exhaustive, and we're certainly not trying to suggest that there's any 'one-size-fits-all' approach, as every track and situation is different, of course. Rather, we're demonstrating a variety of ways in which you can bridge the gaps between separate sections in your tracks. What's more, these techniques aren't at all genre-specific, so you can follow along and brush up on your arrangements no matter what style of music you happen to be producing.

Each technique is accompanied by a video tutorial that shows it off in full detail – all videos can be downloaded from **vault.computermusic.co.uk**, found on the cover DVD with the print edition, or viewed inside the feature itself in the Apple Newsstand edition.

01> Crashes and reverse cymbals

The simple cymbal on beat 1 is a staple of all musical genres. It can be likened to a musical 'capital letter' of sorts, signifying the start of a fresh musical sentence, generally placed at the start of an eight- or 16-bar section. While it might seem like an easy addition to an arrangement, it can actually be quite a tricky one to get right, so it's often worth auditioning several samples in the context of the track to find the right one. Careful EQ, filtering, delay and reverb will help blend a crash into a mix.

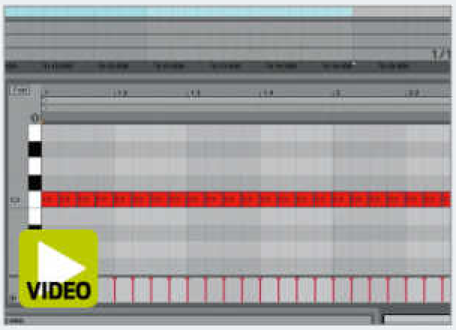
Dance genres will also often make use of a reverse cymbal swell at the end of an eight- or 16-bar section. You can either use a reversed copy of your main crash or a different sample for contrast. For a more floaty effect, try muting the forward cymbal, then carrying over the reverse cymbal with reverb or delay. We show you exactly how in the video.

02> Drum fills & rolls

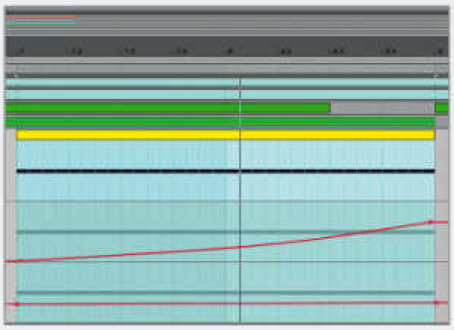
A drum fill is a short rhythmic variation used to 'fill' a gap before a new musical phrase is introduced. It's a tried-and-tested way to provide a burst of excitement, letting the listener know that "something" is coming. In electronic music, you can repeat drum hits as you see fit, but it can be a challenge to program realistic live drum fills in software. Drum ROMplers such as EZdrummer 2 and Addictive Drums 2 include preset MIDI drum fills to spark transitional inspiration.

03> The classic 16th-note snare roll

Electronic music aficionados will recognise the classic 16th-note snare roll, but programming a smooth and effective transition using one isn't always as easy as it sounds. Here's one way to do it…

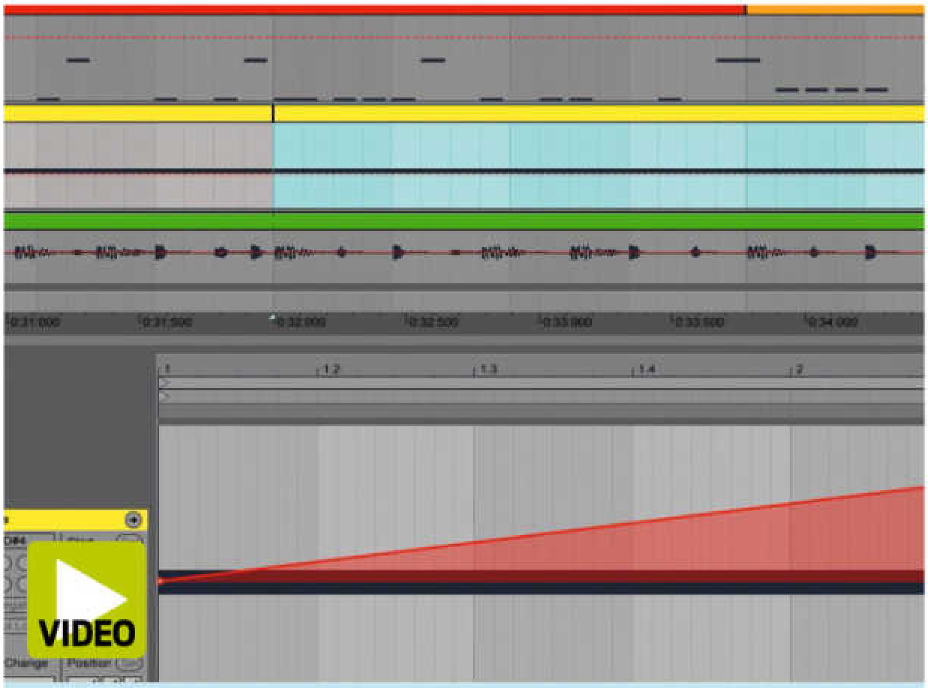


1 > First, we program a 16th-note snare pattern over the final two bars of an intro section that leads into a bass drop. By loading our snare hit into Live's Simpler, we can automate the sampler's parameters for control over the roll's progression, giving it more life and dynamic movement.



2 > We set Simpler's amplitude **Decay** to around halfway and its **Sustain** to minimum. Automating the Sustain to rise over the two-bar duration opens out the rolling snares gradually. Some volume automation also introduces the roll, but this doesn't always need to be a radical shift – often an increase of just a couple of decibels is enough.

3 > Our roll already sounds good, but it can be spiced up in other ways, too, for more individuality. Try automating the Simpler's **Pitch** control for a distinctive pitching up or down effect. Introducing saturation over time can also give snare fills a bit of extra beef; and subtle flanging, phasing or chorus can be mixed in as well, for a touch of weirdness.



04> Selective muting

One of the easiest and most effective ways to transition between two sections of a track is to create rests and pauses by muting certain elements of the arrangement. The ear becomes accustomed to repetition over time, so by removing a repetitive element for a split second using simple editing, you can create dynamics and interest.

This selective muting technique can be used in many different ways, as shown in the accompanying video: you can mute a single drum element – such as a kick or snare – to lead into another track section; or entire track groups, such as drums or bass, can be taken out of the arrangement then dropped back in. One particularly great trick (with origins in hip-hop) is muting drums or whole tracks for the very first beat or bar of a new track section instead of the last. You can also mute everything for just a few beats, and drop a new track element such as a vocal snippet, cool sample or bass stab into the created 'empty' space.

05> Pitch variation

Most of our transition techniques are based on rhythmic ideas or production techniques, but a change in the notes of a bassline, melody or chords can be enough to pique the listener's interest between sections. Check out the video for some note-changing and pitchbending ideas.

06> Glitch-style edits with Eurydice CM

If you're really stuck for a way to transition between track sections, take a look at what's already present in the arrangement. Chances are, you can render one or more elements to a new channel and apply some kind of crazy plugin or audio process for some interesting glitch-style edits. One particularly useful tool in this scenario is Inear Display's Eurydice CM, part of **cm** Plugins. Its four modules – Buffer, Bitcrusher, Filter and Tape Delay – can be used individually or combined for some eccentric edits and transitional effects. It can be a wild beast to tame, though, so it's a good idea to tweak its controls live and record the results onto a new audio track to chop up later.

07> Delay on a return

Sends and returns offer a speedy way to add variations between track sections, especially when used with delay or reverb plugins – and it's simple to render these tracks out for finer editing.

1 > This tune moves between two very different sections and needs a transitional effect to smooth the join. We could add entirely new FX sounds to bridge the gap, but instead we'll add a return track with a delay plugin and automate the vocal's send level.

2 > A suitable delay preset is selected to provide rhythmic floatiness and interest (with the delay set 100% wet). Basic delay plugins aren't always your best bet here – reach for wackier ones with distinctive saturation, filtering or pitch modulation to add a dose of crazy character.

3 > Our delay signal carries the track through to the second section, and the effect fits perfectly because it's been created from the vocal that's used in the track. We record the return as audio before muting that channel. The audio signal can now be chopped, processed and manipulated.

08> Filtered noise

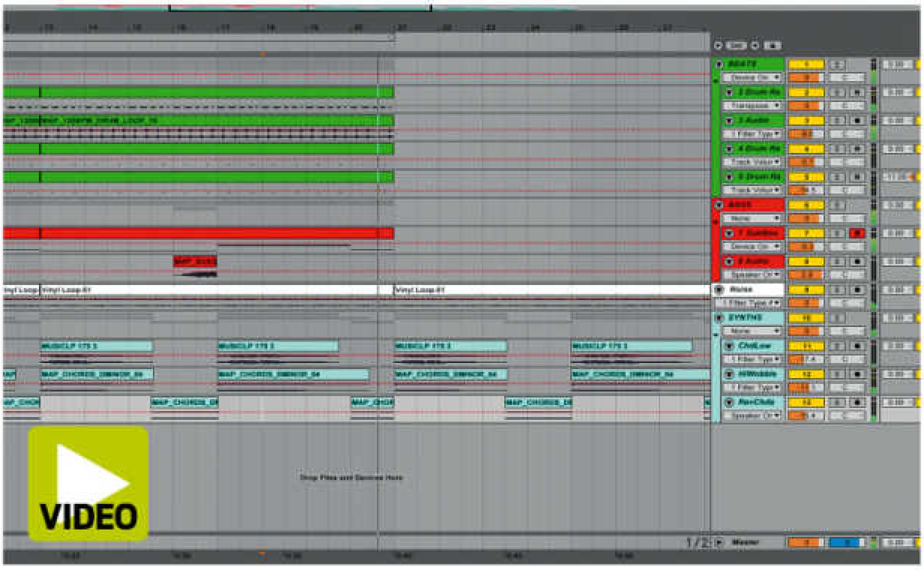
We couldn't discuss transition effects without covering the technique heard in a thousand dance tracks: good old filtered white noise. Simply take a section of white noise (either as audio or from a synth), apply resonant low-pass or band-pass filtering, then add spatial effects to taste. The trick with this kind of effect is either to keep it very subtle, or to customise the rising whoosh effect with plenty of saturation and metallic-style effects, such as extreme flanging, phasers or chorus. In the video, we demonstrate a couple of ways in which it can be done.

09> Anacrusis

An anacrusis is a series of notes that leads into the first downbeat of a bar, so the melody effectively starts a beat or two early. Robert Miles' *Children* and Tina Turner's *I Don't Wanna Lose You* are two examples. In our musical example, we have a verse section that goes into a main chorus drop, at which point a lead riff enters. To emphasise the anacrusis, we mute all the track elements for two beats just before the first downbeat of the chorus, to allow the extra notes to be heard on their own. This can be combined with a drum fill, simple beat roll, reverse cymbal or any other transitional trick – whatever's best for the track you're working on!

10> No transition!

Often, the skill in making a track flow is knowing when *not* to add any extra embellishments, effects or fills. More laidback, progressive or melodic styles, especially, benefit from understated drifts between sections, letting the addition or removal of key elements do the talking. Here's a smooth, melodic track – extra drum fills, muted parts, crazy effects or other tricks will actually detract from the relaxed feel of the whole song. The occasional low-key reverse cymbal or white noise swell may be the best transitional element to use when absolutely necessary, but there's still a fine line between subtle swooshes and effects that just distract for the sake of it.



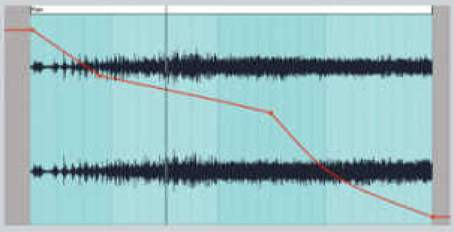
TIPS & TRICKS

KEEP IT CYMBAL, STUPID

In our first video tip (on cymbals), we demonstrate the use of long reverb and delay for a floaty crash effect. This might not suit your style of music, particularly if you want a more choppy, upfront feel. In this case, experiment with a more 'choked' cymbal: shorten your crash sound – using editing or envelopes – to a beat or so long and widen it with a shorter delay or reverb.

MAKE SOME NOISE

While filtered white noise is commonplace, you can get unique and personalised effects by filtering different types of noise. Many synths offer more unusual and distinctive noise waveforms for use as filtered 'whooshes', or you can load up your own field recordings of street noise or other textured sounds for a truly personalised riser effect.



You can get a more bespoke white noise signal using broadband found sounds – this one's an electric fan

BIGGER PICTURE

Creating a transition or fill is often a process of trial and error, which is why you'll see that we often play our creations several times in the videos, adjusting things as we go. When we're happy with the effect, we hit play from a little earlier in the arrangement to audition the sound in the full context of its surroundings. Remember that transitions have to work with the overall flow of your track, not just on their own.

EVERYTHING TO GAIN

Level fades can be used as simple but effective transitional tools. Load Live's Utility device or any other effect with a gain knob onboard over a drum or bass group, then simply automate the main level down leading towards a key section – ideal for smoothly fading into a breakdown, or creating tension before a drop, while keeping your volume fader free from automation so as not to confuse later level adjustments.

Use low- or high-pass filtering across one or more track elements to really reinforce a transition between sections

FILTER FREAK

Filter plugins alter a signal's frequency content, making them a perfect tool for transitions. Sweep a low- or high-pass filter over individual tracks, groups or your master output to reduce the frequency range, creating tension before a key chorus or drop section. Wild sweeping and high resonance settings will give a more extreme DJ-style sound, while subtler applications can be less obvious but no less effective. Filtering works especially well when combined with other effects like fills or rises.

WHERE'S YOUR GRAMMAR?

As mentioned earlier, a crash cymbal is usually used as a musical 'capital letter' on beat 1, marking the beginning of your musical 'sentence'. The listener often expects this, so toy with their expectation. Try moving your cymbal to beat 2, perhaps combined with some choice pausing of other elements; or use white noise or an effect 'down swoosh' rather than an actual cymbal.

BE RUTHLESS

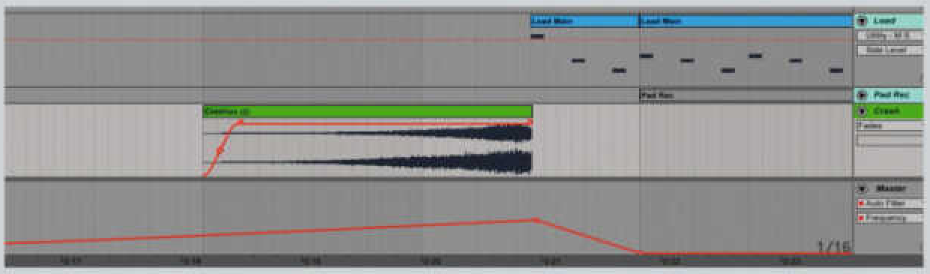
Often, you'll create a transition partway through the writing process, and it'll remain the same for the life of the song. Don't get too attached, though: have a critical listen through and realise when a fill or transition isn't working, then be brutal and pick apart your bridging techniques, going back and adjusting where necessary.

INSERT DELAY

We've discussed the use of delay on a return track, but try using delay or reverb plugins as inserts directly on a sub-group. By automating the Wet/Dry (or Mix) parameter on a delay or reverb, the main dry signal will be lowered as your spatial effect is increased. Automate this control to open the effect right before a drop, then slam the mix back to 100% dry just before it kicks in.

STRENGTH IN NUMBERS

Often, it's not one single technique that glues separate sections of an arrangement together, but the clever combination of multiple transitional tricks. Try pausing your core musical elements when a rise sweeps up, sending a drum fill effect to a twisted delay effect on a return, or applying glitchy processing to a 'lead-in' section of the main melody. cm



Sometimes the true mark of a good transition is the evolution of several track elements at once – not just one

LOOP LAB

From creation to manipulation, loops play a key role in the modern production process. So what does it take to master the art without getting yourself in a twist?

> In music production terms, a loop is a segment of audio that seamlessly cycles round and round. Without repetition, music would be unpredictable and unsatisfying for many listeners. Even the most complicated scores rely on recurring elements to underpin more complex ideas, giving us a structure (and often a rhythm or groove) to hold on to. Electronic music in particular relies upon the repetition of drums, synths and samples, and the design of basic sequencers and drum machines has always been geared up for just that.

As modern music making has shifted over to the home-based producer, so the popularity of audio loops has risen, and commercial sample creators such as Loopmasters, Sample Magic, Vengeance and Samplephonics all provide a rich variety of loops to cater for all tastes and needs. But why use sampled loops?

For starters, they can provide you with ideas or inspiration that you'd probably never come up with yourself. A pre-made drum loop or guitar riff could easily send your composition down a completely different musical path to that on which it started, triggering other tactics for secondary melodies or even a different piece altogether. You could put a live drummer in the thick of your dance tune, or a professional bass player into your score, for example. In extreme cases, entire genres have been built upon the use of sampled loops – just look at what the humble Amen break did for jungle! Looped rhythms can also help you beef up other foreground parts, giving your beats more weight or depth.

So, in order to help you get the most out of your loops, in this tutorial we're going to take a look at how to manipulate, edit and compile the ones you already own, as well as how to create your own from scratch, to build a bespoke loop library for use in future Live projects.

As well as Ableton Live 9, we'll be using various **cm** Plugins, which you can download from **vault.computermusic.co.uk**.

To loop or not to loop...

In 2010, Steve Angello hit the top of the Beatport Chart with *Knas*, a basic house workout comprising a 4/4 beat over a sliding electro bass riff. Nothing memorable, then... except for the uproar it triggered across electronic music forums. It seems one of the world's most famous producers had simply thrown in a bass loop straight out of a Vengeance sample pack, placed a kick over it, and scored a chart-topping hit.

The debate has been ongoing ever since the birth of sampling itself: should you use other peoples' loops in your tunes, or should you painstakingly program each cycle from scratch using your own sounds? There's no clear-cut answer. While simply chopping a chunk out of a commercial song, blatantly looping it up and bagging a hit record could bring down a barrage of copyright lawyers, there are a ton of high-quality, copyright-free loop packs available for unlimited use in your music. Snobs may look down their nose at the practice, but if a loop fits the bill, then there's no harm in dropping it into your track. Mr. Angello certainly wouldn't argue.

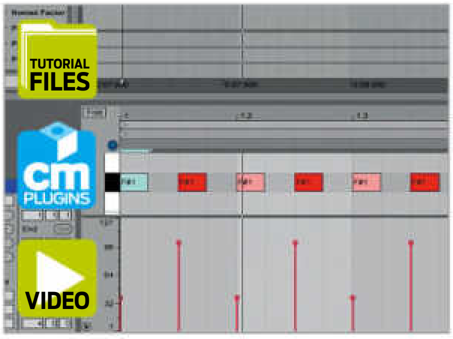
So, distinctive riffs and melodies may easily be spotted, but what about more generic loops and patterns made from one-shots? Modern productions are often densely-layered, complicated affairs, so the task of manually arranging single hits into full tracks is too daunting for some. A compromise would be to combine one-shots and loops in a creative way: piece together your core ideas from synths and sampled hits, then back them up with more subtle 'filler' loops.

In spite of this debate, creating your own custom 'loop' folder is a rewarding task – when you haven't got enough time to work on a full track, it's fun and productive to build fodder for your samples folder, and we'll show you how. Take note, Steve!

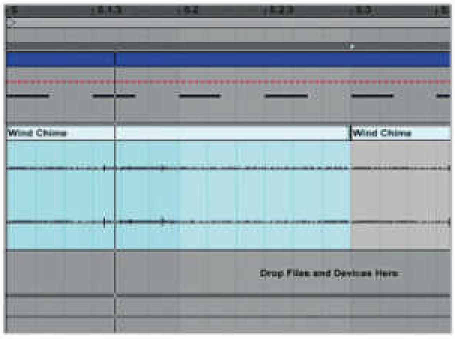
> "The debate has been ongoing ever since the birth of sampling"

> Step by step

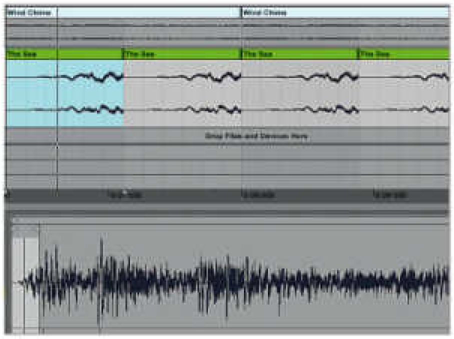
1. Build a drum loop from one-shot samples



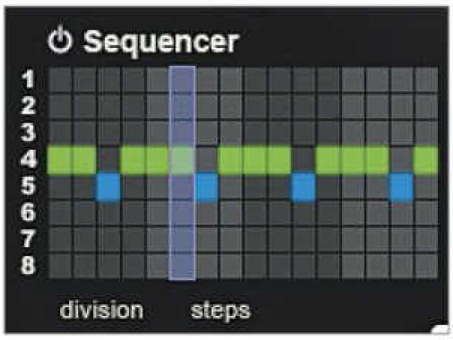
1 > We're going to put together an unusual hi-hat loop using a variety of sound sources. We'll start with a synthetic closed hat pattern. Start a new project at **135bpm**, then add Linplug's CM-505 on a new Instrument/MIDI track. Create a one-bar MIDI region and fill it with eighth-notes on **F♯** (CM-505's closed hi-hat). Accentuate the offbeats by lowering the **Velocities** of notes 1, 3, 5 and 7 to about **27**.



2 > Real-world recordings can add depth and character to a loop. Drop **Wind Chime.wav** onto a new audio track, and trim it down to leave only its first two beats. Duplicate this once, and observe how the 'live' rhythm combines with CM-505's robotic beat as the loop cycles round. Wolfram CM's **CM Every Now and Then** preset adds some subtle spaciness.



3 > Let's add some more recorded texture. Drop **The Sea.wav** onto a new audio track and turn it down by **15dB**, bringing it into line with our other sounds. We'll only use the first beat of this audio, so trim it down and duplicate it four times to fill the bar. This gives us a ducking noise bed under our other elements. Again, we add Wolfram CM for width, this time loading the **Stereo-izer** preset.



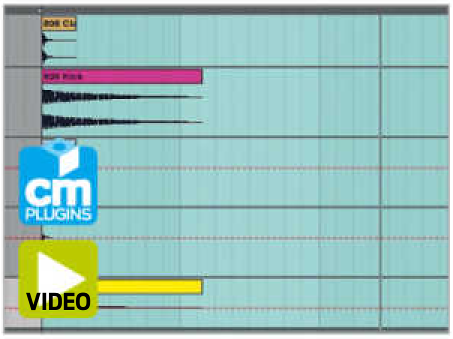
4 > Now we'll add a quirky running percussion part with Loomer Cumulus. Add an instance onto a new MIDI track, then open up the factory library's **Machine Gun** preset. This patch uses Cumulus' sequencer to trigger loop slices over a one-bar period – we rearrange the sequencer steps (by clicking to fill in a step) as shown.

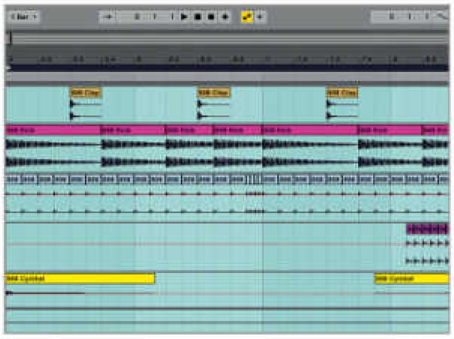
5 > The Cumulus channel contains excessive low-end rumble that could easily clash with other elements when placed in a track, so we resolve this by adding IIEQ Pro CM and engaging its **HPF12** (high-pass filter) at around **205Hz**. As the chime and sea recordings also have wild low-mid and sub content, we then add IIEQ Pro CM to these too, both with a **HPF12** rolloff at approximately **500Hz**.

6 > To finish, we add ReverberateCM as an insert on the master, load the **Smallroom Blumlein02** preset, and bring the **Dry/Wet** down to **0:-5.0dB** for subtle room realism. Kuassa PreMix CM's **Gain** is then pushed to around **+16dB** and its **Output** is backed off to **-10dB** for gluing crunch. Finally, turn PreMix CM's **Low** to **9 o'clock** and the **High** to around **2 o'clock** for some gentle EQ tilting. Now export the bar-long loop into your samples folder!

> Step by step

2. Putting together a custom ‘construction kit’ from scratch





1 > Commercial sample packs often contain ‘construction kits’: collections of interchangeable loops that can be combined to build up a full track. We’re going to create our own construction kit using **cm** synths and samples. Open a fresh Live project, set the tempo to **140bpm** and drop each sample from the **808 Hits** folder onto its own audio track. Set up a four-bar loop cycle to begin.

2 > We arrange our 808 hits in a simple trap-style arrangement. The clap sits on every third beat with the kick beating a plodding rhythm as above. The closed hi-hats play on continuous eighth-notes, then speed up to 32nd and 16th repetitions on beats eight and 16, alongside a snare roll. A cymbal accentuates the start of our loop, with a copied and reversed cymbal adding a final touch at the end.

3 > Our 808 kit sounds thin and weedy, so let’s heat it up with some saturation. We load Kuassa’s PreMix CM onto each individual track and use liberal amounts of **Gain** to add lots more punch and character, backing off the **Output** when necessary to keep levels under control. We’ve got much beefier, crunk-style beats now!

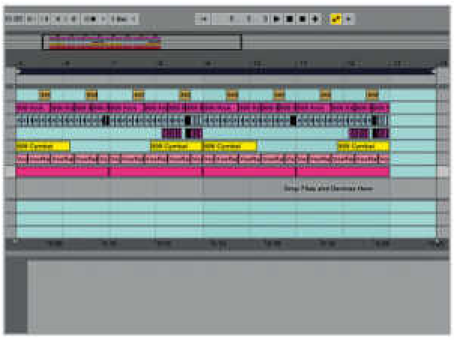
4 > Our hits are extremely dry, so let’s add two separate instances of Reverberate CM onto Return channels – load the first with the **Drum-01-44100Hz** preset, the second with the **Hallway Blumlein01** preset, and turn the **Dry/Wet** amount to fully **Wet** on both. Send the clap and cymbal to the first, longer reverb; and the closed hat and snare roll to the shorter room.

5 > Let’s get a low-pitched vocal loop going. Drop **VoxRain.wav** onto a new track, place it on the first beat, turn it down by **8dB** and pitch it down by **4 semitones**. Now cut the vocal up into phrases and repeat sections to create a repetitive hook. Put Wolfram CM’s **Smeared Distortion** preset on the vocal channel to add character, then apply maximum send amounts to the two reverbs for width.

POWER TIP

>Chopping off tails

Although here we’ve chosen to export the main looping section of our construction kit, it’s worth noting that we can also export out the beginning or end sections with the reverb or delay tails. When we later construct our track using the kit elements, we can add this natural overlap to any intro or breakdown sections so our sounds aren’t ‘chopped off’. Many loops don’t feature these tails, however, so you may want to send your loop to a delay or reverb return before dropping to a break.



6 > It’s time for a lead line. Open SynthMaster CM on a new MIDI track, load the **Trap Lead** preset (found in the **Tutorial Files** folder) and turn the track down by **10dB**. The patch’s squeaky pitch modulation is controlled by Mod Envelope 1, found in the Modulation tab. We program in a simple two-bar riff as MIDI, then duplicate the region and transpose bar three and four’s notes down an octave for variation.

7 > Our composition is finished, so we can now compile a ‘construction kit’ from its elements. As we have endings that may be cut off abruptly, let’s duplicate all the tracks over another four bars, then set the project’s loop markers around all eight bars, plus an extra bar to catch delay and reverb tails – ie, nine bars in total. We now solo each track in turn and render them into a folder, ensuring our reverb returns are part of the resulting audio.

8 > We could use our loops as they are, but we’re going to edit them a bit further, so they loop perfectly over four bars. Import the rendered loops onto separate tracks in a fresh 140bpm project, then set the loop markers around bars 5-8. We can now export our tracks as four-bar loops with their reverb and delay tails intact. There we have it – a perfectly looping construction kit, ideal for quick track-starting!

Five top tools for loop creation

Propellerhead ReCycle

ReCycle is now a true veteran of the loop-slicing game. Simply load an audio file into the standalone editor, place slice markers at the points you want to slice, and then export out a REX/RX2 file. Each slice can now be pitched and rearranged in your REX-compatible sampler plugin.

Sugar Bytes Effectrix

This creative mangler, once loaded as an insert over a track, allows 14 different effects to be combined over a tempo-synced grid. Chopping, stuttering, looping, filtering, pitchbending, modulating – piecing together crazy patterns of multiple effects has never been easier.

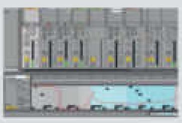
Native Instruments Maschine

A drum pad-based hardware MIDI controller and software browser-cum-step sequencer combine to form the ultimate environment for loop creation, either in standalone mode or within your host. Navigate your hard drive on the fly and throw together inspiring loops quicker than you can say "MPC".

Spectrasonics Stylus RMX

Whilst it may be showing its age, Spectrasonics' REX manipulator still holds its own thanks to its immense sound library and creative potential. Its 'Chaos Designer' is a standout feature, letting you randomly reshuffle, reverse and glitch up your loops beyond recognition. Time for RMX 2, guys?

Ableton Live 9

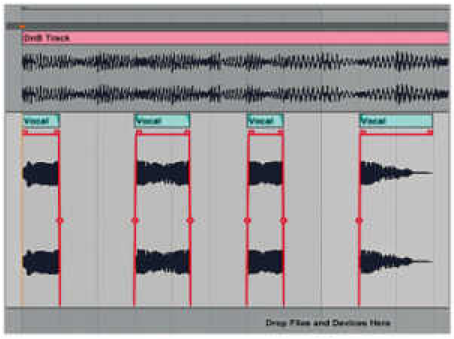


Then, of course, there's Live itself. Whilst most DAWs are based solely upon the traditional left-to-right sequencing design, Live has been built with the cyclical musician in mind. The Session View lets you trigger audio and MIDI over a loop-based sketchpad, then transfer those ideas to the Arrangement View.

> Step by step

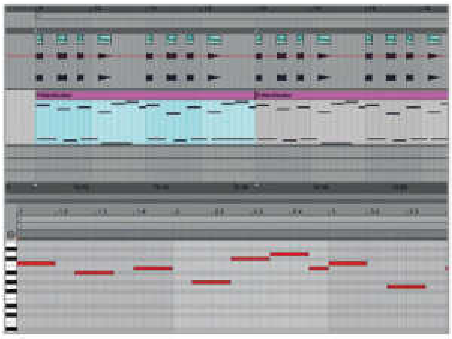
3. Creating a choppy vocal from a single ad lib

1 > Let's create a quirky vocal loop using one small vocal phrase. Set Live's project tempo to **174bpm** and drag **DnB Track.wav** onto a new audio track. Next, drop **Vocal.wav** onto a second track to start at the same time as the first. Set the loop markers around the first two bars.



2 > Set Live's grid to **1/8** and cut the vocal into four sections at the grid markers using **Ctrl/Cmd-E**. Pull the vocal segments apart so that there are two eighth-note spaces between each slice – you may need to apply volume fades at the slice points to prevent clicks.

3 > For variation, let's transpose the second vocal segment down **-7** semitones and the fourth segment **-12** semitones. We want an interesting variation of effects, so load up Sugar Bytes' Artillery2 CM Edition as an insert on the vocal track and open its **Super FX Keyboard** preset. Create a new **MIDI Track** and set its output to the vocal track's input so we can trigger the effects with our controller keyboard (or via the piano roll).



4 > Artillery2 CM's resonant low-pass filter and stuttering looper modules will give our dry vocal some quirky character, so we spend some time programming notes on the MIDI track to really emphasise the chopped-up loop. You can either draw in your own pattern here, or import **FilterStutter.mid** onto the MIDI track to see how we did it.

5 > The effected vocal track's levels are jumping around, so let's tame them with Hornet's Fat-FET compressor. We use a **Threshold** of around **-35dB**, a **Ratio** of **8:1** and a **Release** of around **69ms** to even out the signal's volume. An **Output** increase of around **4dB** brings its level back up in the mix.

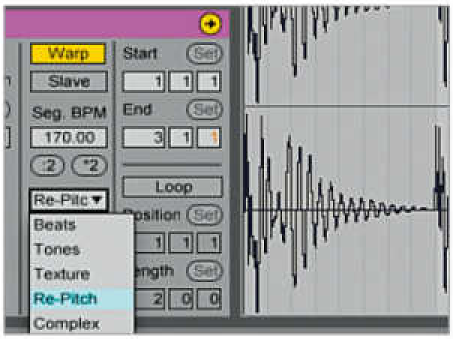
6 > To take the dry edge off the chopped vocal, we'll finish with some stereo delay. Add KR-Delay CM after the compressor and set its **Mode** to **Ping-Pong**. A **Dry/Wet** amount of **20%**, Delay time of around **117ms** and roughly **12.5% Feedback** give subtle widening to our loop. We finish off by bouncing our vocal channel out as a two-bar audio file (**Vocal Final.wav**).

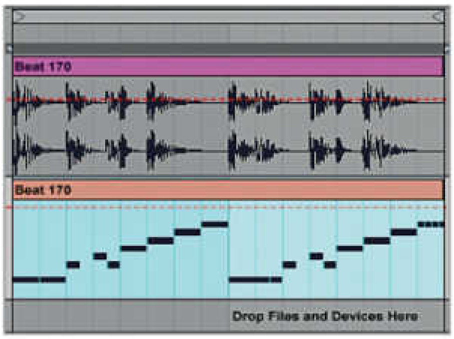
> Step by step

4. Getting loops to work at different tempos

1 > Many producers stick to using audio loops that are similar in speed to the styles of music they create, but this rules out many great sources of loops. Here we're going to show you several different methods for manipulating a drum loop's tempo, so set your project to **170bpm** and load **Beat 170.wav** onto a new audio track.

2 > Timestretching will alter the length of an audio file while keeping its original pitch intact. Live detects the loop's original tempo as **170bpm** and automatically stretches it if we change the master tempo. This method is great for short BPM alterations but can sound unnatural at extreme values. Change the project tempo to **127bpm** and check out those artifacts!





3 > Drums rely on transient detail, but even Live's excellent timestretching algorithms can cause glitching or loss of attack. Simple repitching is a good way to get around this, since your loop's pitch will be altered in conjunction with its tempo (like pitching a sound up or down in a sampler). You can achieve this in Live with the **Repitch** algorithm.

4 > What if you want to drastically alter your loop's BPM *and* keep its pitch? Live's 'Slice to MIDI' feature can help. **Right-click** the loop and select **Slice To New MIDI Track**. Live chops up each transient and puts each slice in its own Simpler on a new MIDI track. We can now open up the resulting MIDI region and rearrange the hits.

Slice or stretch?

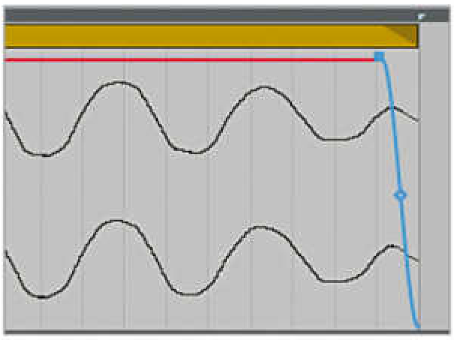
So when should you timestretch a loop to fit, and when should you slice it up to reprogram each hit as MIDI? It depends on the length and type of loop.

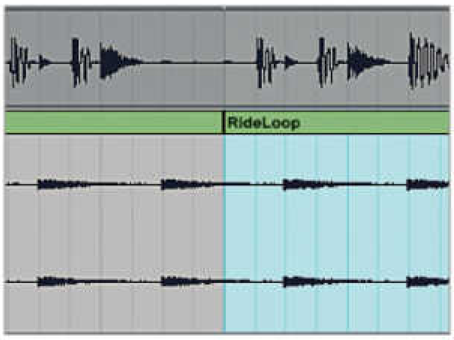
If you're on a tight schedule or need to stretch longer/multiple regions, then timestretching will probably be the solution, using Live's powerful elastic audio technology. This method will also keep the loop's original composition and flavour intact.

The downside is that a loop's tempo can only be pushed or pulled so far away from its original tempo. Slicing up a loop and mapping each hit to separate pads in a drum sampler means that each segment is played back at its original speed, no matter what the project tempo is – a great solution when you want to pitch, envelope and reprogram each hit in a far more precise way via MIDI. This isn't magic, though, as slicing a 170bpm loop up and replaying it at 115bpm will still leave gaps in between each hit; but some tuning and a touch of reverb can fill in smaller gaps with no stretching artifacts.

> Step by step

5. Editing drum loops for perfect looping and consistency





1 > It's good practice to keep your loops neat and consistent in volume for auditioning purposes, so we're going to tidy up a few loops before exporting them to our main Samples folder. Set Live's **Tempo** to **140bpm**, import **ElectroBeat.wav** and **RideLoop.wav** onto separate audio tracks, then loop the audio.

2 > When a digital audio file's region ends in the middle of a wave cycle, it causes an unwanted click at that crossing point. **Solo** KickLoop.wav and listen out for the click at the end as its gets cut short. We can remove this click using a tiny volume fade at the end. **Right-click** the audio clip and select **Show Fades**, then slightly drag the right side's volume node to the left to apply a tiny fade and remove the click.

3 > Our ride loop is very quiet compared to the first beat, so raise its level by **10dB**. A small gap can be heard at the end of the loop, so tidy this up by doubling over the first bar of the ride region. Now the loops are neat and level, we can export them both into our Drum Loops folder. Name them appropriately – we've chosen the names **DrumLoop Electro 909 140** and **RideLoop 909 140**. **cm**

DOWNLOAD
Get the tutorial files to follow along on your PC/Mac at **vault.computermusic.co.uk**

PHAT CHANCE

Electronic music needn't be predictable and repetitive. Using Ableton Live's randomisation features, you can program the machine to be uniquely creative every time

> **While repetition is a fundamental feature of contemporary electronic music, one of the biggest challenges facing producers is the avoidance of monotony at both the macro and micro levels.**

We're always looking for new ways to imbue our sounds with life and movement; for example, on the macro level, we might add live instruments to otherwise tightly programmed pieces, bringing back that 'human element'. And on the micro level, we often turn to less predictable, non-linear analogue emulation plugins to warm up 'cold', 'digital' sounds.

Such techniques and tools can make our music sound alive and organic as opposed to artificial and robotic, but we can also harness unpredictability to break away from the inherent compositional tendencies that subconsciously guide us all, to generate unique ideas and sounds from scratch.

Using the elements of chance and randomisation, we can get our computers to do the creative legwork for us within a specified range of variables and rules. Ableton Live 9, used in conjunction with Max For Live's prefab devices, constitutes a powerful set of tools for MIDI programming, and includes some specialist utilities for randomising parameters within Live. We're talking chaos-minded MIDI effects devices such as Random and MIDI LFO, as well as the 'micro' Random parameters on some of Live's devices (Impulse and Collision, for example). It's fair to say that Live has more onboard, readily usable randomisation functions than any other DAW.

In this series of walkthroughs, we'll explore the use of Live's randomisation features in a range of scenarios in order to familiarise you with various generative music techniques and concepts that can be used both in studio production and live performance. From generating chords, melodies and patterns to humanising electronic drum sounds by randomising parameters, or even composing complete generative ambient drone pieces, these techniques are an introduction to the crazy world of random-based MIDI programming.

We're going to take an introductory look at the use of random parameters in production, but if this gets your juices flowing, you can delve further into the philosophy and art of generative music by hearing Brian Eno's take on the form he popularised at **bit.ly/15TiCrq**.

Get more from Max

Max For Live comes with a great collection of preconstructed instruments, effects and tools, but there's also a sizeable community of artists and developers for the platform, building and sharing an extraordinary range of devices.

The Max For Live Essentials bundle of instruments and effects features three devices offering notably good randomisation functions. First is LFO MIDI, which we're using in most of our walkthroughs here. Although ostensibly just an LFO that can be used to modulate pretty much any parameter within Live, one of the waveforms in its Shape menu is a random output generator. Oh yes! Then there's Buffer Shuffler 2.0 and Mono Sequencer, both of which sport randomisation options for manipulating incoming audio signals and outputting random MIDI sequences.

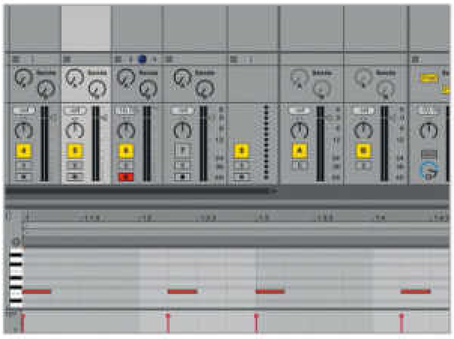
Maxforlive.com hosts a growing library of community-made M4L devices (1418 at the time of writing), including plenty with nifty randomisation features, all free to download. Among them is K-Devices EXT, which enables you to modulate parameters via a step sequencer with the ability to change the pattern randomly on each bar.

Of course, the real beauty of Max For Live – for those with enough time on their hands, at least – is the possibility of building your own randomising devices with it.

> Step by step 1. Randomising a sample's start point

1 > Create a MIDI track in Live, add a **Simpler** device and load a sample – we're using **Atmos27.wav** from the **Tutorial Files** folder. We're going to randomly modulate the sample's start point to generate interesting tone and amplitude variations. This technique works really well with long guitar loops, but try it with anything – even full tracks!

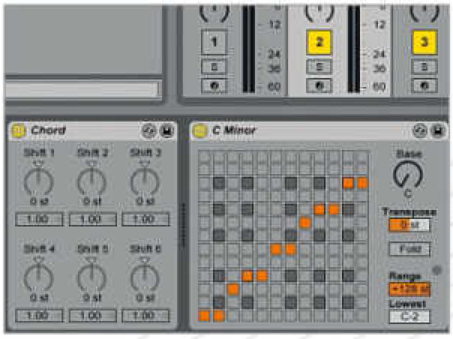
2 > Set the **Release** time to **2.00s**, **Attack** to **0.5ms** and **Length** to **20%**. Turning on **Snap** mode prevents unwanted clicks by making the loop boundaries snap to zero-crossing points. You can also change the **Spread** parameter to around **15%** to add some stereo image. Currently, whenever you trigger a note, the sample plays back from the start.



3 > Create a new MIDI clip and program a simple pattern triggering just one note. We've gone for a minimal house staple that works particularly well with a four-on-the-floor kick pattern. We're triggering with the note C3, which plays the sample at its original pitch.

4 > So far, so repetitive – let's use randomisation to add movement. Insert an LFO MIDI device. Set the **Shape** to **Random** and **Depth** to **58%**, then add some **Jitter** and **Map** it to Simpler's **Start** parameter. With random starting points on every note, we get constant tonal variation. In **Result.wav** you can hear what happened in our version.

> Step by step 2. Randomly generating chords and melodies with Chord MIDI and LFO MIDI



1 > Drag Live's **Chord MIDI** Effect from the Browser onto a MIDI track. Place a **Scale MIDI** effect after it, and set it to the **C Minor** preset. You can use any polyphonic synth or sampler for this – we're calling on the Live Instrument Rack preset called **Dance Dot Org**, which is an organ sound.

2 > Now we're going to use Max For Live's LFO MIDI device to control the Chord plugin, creating random chords and melodies. Drop **LFO MIDI** onto the track, set the **Shape** to **Random**, **Depth** to **10%**, **Offset** to **0.8**, and **Map** it to control **Shift 1** on the Chord plugin. This randomly inserts high notes, between +17 and +36 semitones above the input note, into any chords and melodies on that track.

3 > Now lets generate some randomness in the lower octaves. Drop another LFO MIDI on the same MIDI track and set its Shape to **Random**. Change the **Offset** to **-0.40** and **Depth** to **6%**. Set a slow **Rate** of **0.24Hz** to randomise our low octave notes less frequently than the higher ones. Ours is available to hear in **Chords.wav**.

Impulse's randomisation features

When Ableton introduced the Drum Rack format to Live in 2007, the huge flexibility it brought to the table – including the ability to host 127 complete instrument/effects chains rather than seven samples – convinced most Live users to pretty much abandon its less powerful percussion-orientated predecessor Impulse. But despite its obvious limitations, Live's original multi-pad sample player boasts several unique features that shouldn't be overlooked, especially when it comes to drum programming.

Abilities like timestretching each sample and transposing the whole kit up and down are useful enough for sound design purposes, but what makes Impulse stand out is the option to easily randomise the Transpose, Pan and filter Resonance for each sample slot.

Emulate the imperfection of a live drum recording with subtle pitch randomisation, and add stereo movement by randomising panning. Used subtly, filter resonance randomisation can also be helpful in adding motion to drum samples, or, indeed, any other sounds. Impulse doesn't have velocity randomisation, but prefix it with a Velocity MIDI effect with the Random range turned up to vary the strength of your MIDI notes and there you have it. This also enables you to randomise Impulse's Stretch parameter, since it can be modulated by velocity.

Below we'll use Impulse with Live's MIDI effects to create a generative hi-hat pattern that changes over time. This technique can be used with any percussive patterns and melodic sequences.

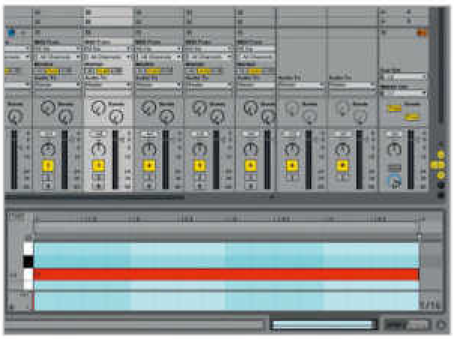
Once you've got this technique mastered, you can create multiple instances of the same chain and use them with different sounds and settings to build complete drum tracks.

Don't dismiss Impulse as old hat – it's got plenty of cool randomisation capabilities

> Step by step

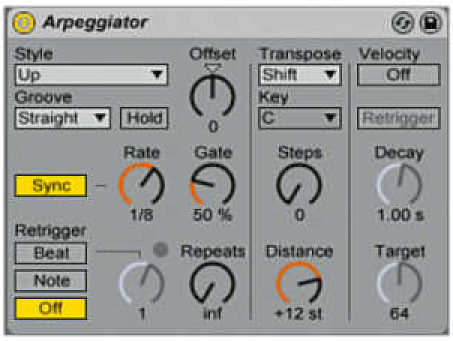
3. Randomising trap hi-hats in Impulse

1 > Start a new Live set and set the tempo to around **70bpm** – a standard tempo for trap music. Create a new MIDI track and load an Impulse device onto it. Drag **CH.wav** onto the first sample slot. Now we have a hi-hat sample triggering on note C3.



2 > We've made a backing track to accompany the hi-hat so that you can hear how it sounds in context – drop **Beat1.wav** onto an audio track. We'll generate our hi-hat patterns using just one note. Create a one-bar MIDI clip on the hi-hat track and fill it with a C3 note.

3 > To transform our long MIDI note into a rhythmic pattern, we'll use Live's Arpeggiator MIDI effect – drop one in before Impulse. If you play the MIDI clip (or a C3 note on your MIDI keyboard), the default settings on Arpeggiator will generate eighth-notes synced to the project tempo. (**Beat W Hihat1.wav**)



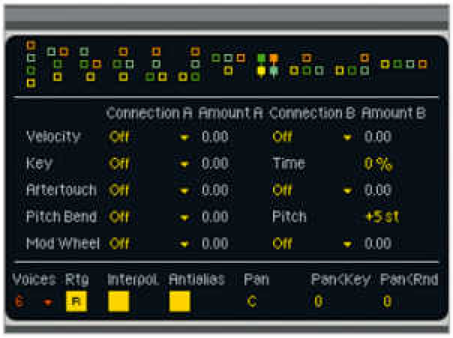
4 > Let's bring some life to the hi-hats with Impulse's randomisation features. To randomise the Pan position, set the **Random** parameter under the **Pan** knob to **40%**. This randomly shifts the pan position with every incoming note, creating a more dynamic sound. Now set the **Random** parameter under the **Transpose** dial to around **35%**. This introduces subtle changes in pitch.

5 > We've applied random movement to the pitch and pan of our hi-hat sound, and the next step is to introduce some velocity variation, which simply controls the volume of the sounds generated by each note. Put a **Velocity** device after the Arpeggiator and set the **Random** factor to **25%**. Now the velocity of the notes will be randomly adjusted up or down.

6 > Finally, let's use randomisation to create an unpredictably evolving hi-hat pattern. Drag Max For Live's LFO MIDI device onto the hi-hat track. Set the wave **Shape** to **Random**, the **Depth** to **6%** and the **Offset** to **0.10**. Change the **Time Mode** from Freq to tempo **Sync**, and the **Rate** to **1/2**, and **Map** the LFO to the Arpeggiator's **Rate** parameter. (Audio: **Beat W Hihat2.wav**)

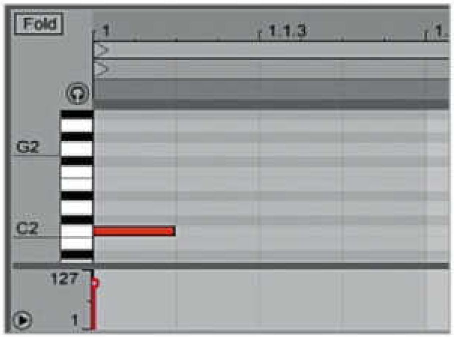
> Step by step

4. Creating generative ambient/drone sounds with Operator



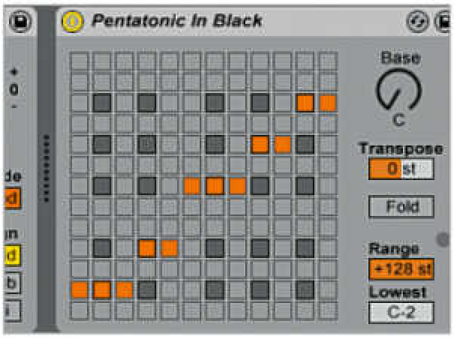
1 > Load an **Operator** synth into a new MIDI track in Live. Change the routing algorithm to two oscillators and two modulators by selecting the fourth algorithm (block diagram) from the right. In this mode, Oscillator B modulates Oscillator A, and Oscillator D modulates Oscillator C, giving us a good starting point for ambient drones. While you're at it, change the **Voices** parameter to **20**.

2 > Select Oscillator A, and set its Envelope **Attack** to **5s** and **Release** to **60s**. Sine waves are a good bet for drone sounds using an FM synth like Operator, so set Oscillator A's **Wave** to **Sine**. Do the same for Oscillator C, then set its **Attack** and **Release** to similar settings to those of Oscillator A, its **Coarse** tuning to **5** and **Level** to **-15**.

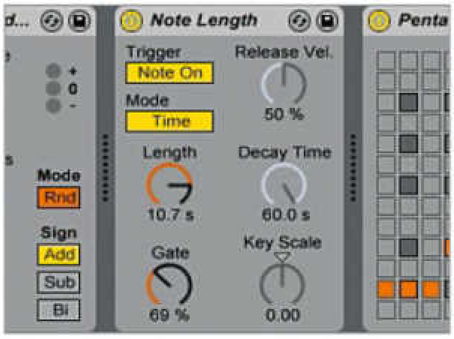


3 > Now we have two Sine wave generators – one in the low frequency range and one higher up. Let's now program a generative note cluster. Create a one-bar MIDI clip and draw a short C2 note at the beginning of it.

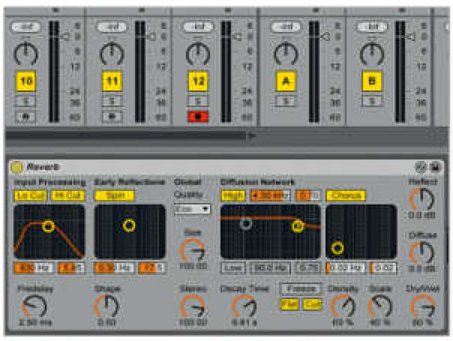
4 > Let's shift this C2 note to a random pitch. Insert a **Random MIDI** plugin before Operator – it generates a random MIDI note pitch whenever it receives input. Raise the **Chance** parameter to **70%** and **Choices** to **24**. Now, 70% of the time, a random note up to 24 semitones higher than the input note will be generated.



5 > This is quite cool, but the random notes don't belong to any particular scale. To quantise the incoming notes to a scale, insert the **Scale MIDI** Effect between Random MIDI and Operator and load a preset – we've gone for **Pentatonic in Black**. Now when you play the clip, the result will be much more musical.



6 > Now we have our 'generative engine', but we need to stack the notes up over time to make an ambient drone. To do this, insert a **Note Length** plugin between the Random and Scale plugins and set the **Length** to **10s**. This will hold the incoming notes for ten seconds. At this point, you might want to turn the track's level down to avoid clipping the master buss.



7 > For a more evocative sound, insert a **Reverb** after the Operator, set its **Decay Time** to **10s** and **Dry/Wet** to **80%**. In the Input Processing section, turn on the **Lo Cut** filter to prevent low frequency rumble. Now we have a suitably spacious base sound for our drone – let's randomly modulate it!

8 > So far we've not used Operator oscillators B and D, which are our FM modulators. Select Oscillator B and set its **Wave** to **Square4**. Insert an LFO MIDI device before the Operator, and set its **Offset** to **-0.20**, **Rate** to **3.00Hz**, **Depth** to **5.00%** and **Smooth** at 9 o'clock. Set the **Type** to **Random** and **Map** it to Oscillator B's **Level** parameter.

9 > Now we use the same technique for Oscillator D. Insert another LFO MIDI before the Operator. Set the **Type** to **Random**, **Depth** to **44.8%**, **Rate** to **1.00Hz** and **Smooth** halfway up. **Map** it to Oscillator D's **Level**. Play back the MIDI clip and enjoy some evolving ambient sounds! Hear ours in **Drone.wav**. **cm**

HOUSE TRACK-BUILDER

Build a dream house track from scratch using Ableton Live and our exclusive free plugins

DOWNLOAD

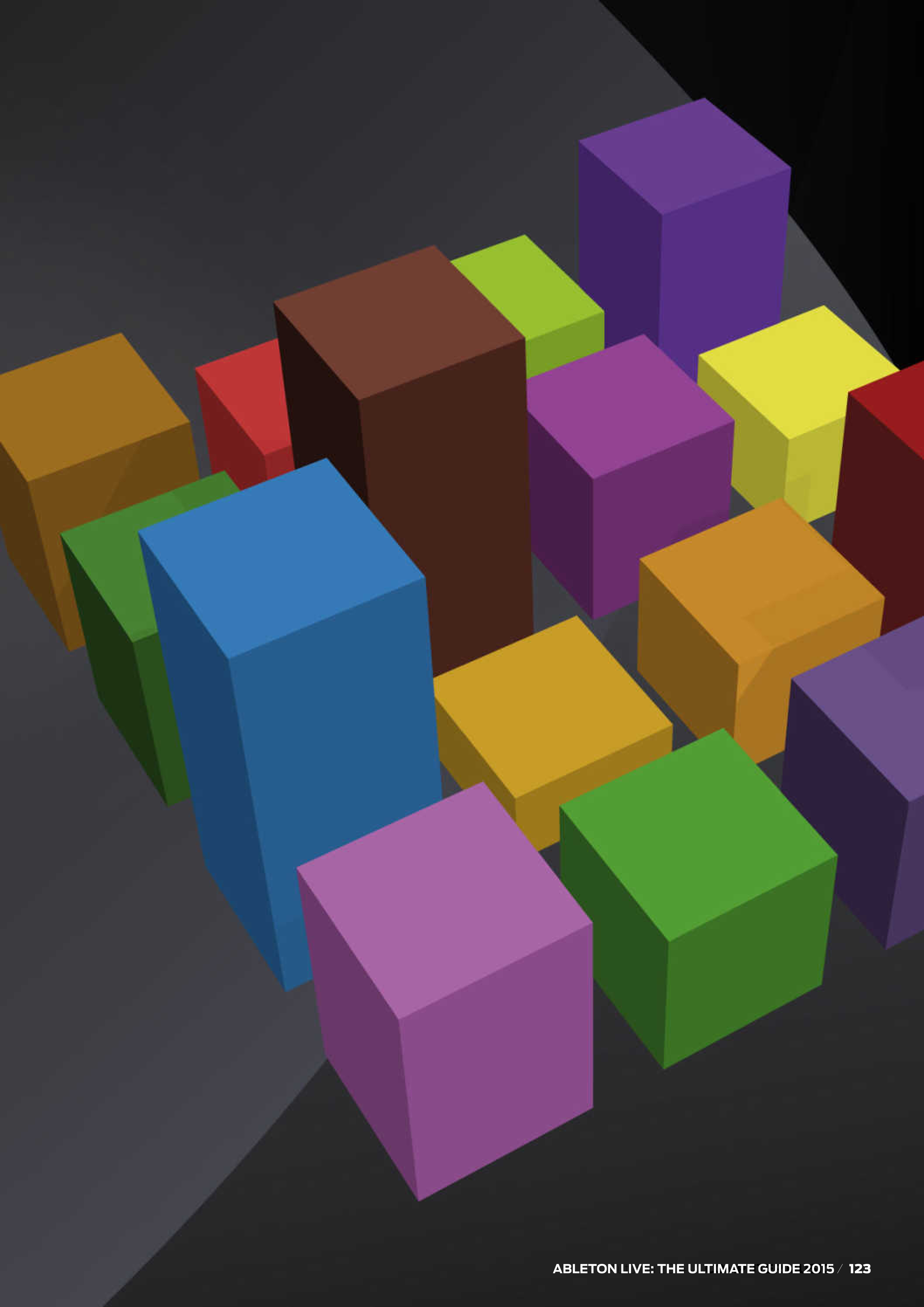
See the videos and get the plugins and files you'll need at **vault.computermusic.co.uk**

> House music is arguably the most influential electronic genre in the history of popular music. Its origins can be traced back to 1980s Chicago, where a handful of DJs including Frankie Knuckles and Ron Hardy championed an eclectic blend of dancefloor-oriented music in the wake of disco; maintaining the soul of disco and pop, while incorporating the calculated repetition of electronic artists like Kraftwerk. This futuristic club sound, and the emergence of affordable music-making hardware from manufacturers like Roland, inspired producers to create their own brand of synthesiser and drum machine-based compositions. The 4/4 kick, synth hooks, samples and disco diva vocals came to epitomise the genre, which gained traction over the years, evolving into the popular electronic form we're familiar with today.

Fast-forward to 2014, and it's evident that house music is more popular – and more versatile – than it's ever been. It spans from the coolest underground club movements to the upper reaches of commercial culture, with mainstream pop acts regularly jumping on the 'house' bandwagon. The 4/4 kick and off-beat hi-hat can be heard everywhere, and it's tougher than ever to compartmentalise 'house'. Countless sub-genres cater for every taste. Deep, progressive, hard, funky, tech, bassline, tribal, Italo, electro, fidget, Dutch, ghetto…

Successful house producers draw from a wide palette of sounds and influences, and we could fill an entire book discussing the range of artists out there today. There's the garage-inspired chart hits of Disclosure, Deadmau5's synth-heavy minimalism, Todd Terje's elaborate synth-disco, the bassy hip-hop vibes of Claude VonStroke and the Dirtybird camp, deep analogue grooves of Jamie Jones, Shadow Child's DnB flavourings, Seth Troxler's Detroit-style retro tech, and countless more. Phew! And that's without touching on the wave of big-room 'EDM' producers – Avicii, Porter Robinson, Nicky Romero et al – that are shaking up dance music across the globe.

A recent trend in house music is the revival of the 'vintage' sounds used in the original Chicago creations – combining retro Roland drum machine sounds, DX7-esque FM basses and analogue synth tones with crusty samples and chopped diva-style vocals. It's a distinctively classic, timeless house sound that a whole new generation of fans and producers alike are discovering for the first time, and over the next few pages, we're going to show you how to achieve this popular sound using Ableton Live and a variety of free instruments and effects. We'll provide you with all the plugins, samples, tricks and techniques you need to craft a vintage-inspired house track. For every walkthrough, there's an accompanying Live 9 project file, as well as all the other files you'll need to make it yourself from scratch. We've also included videos of every aspect of the track's construction, along with a WAV of the finished product, all available to download!



Getting started

We'll be handling every stage of the production process in Ableton Live 9, and the project file from every walkthrough is in the Tutorial Files folder (along with all the necessary audio, MIDI and preset files). To get the files and plugins you need, copy them to your system from the DVD or download them online from **vault. computermusic.co.uk**.

With all that sorted, let's get started building our 124bpm house track from the ground up. AudioRealism's ADM CM hosts a multitude of high-quality drum sounds, so it makes a fine starting point for that most essential house music element: the kick drum. In the first walkthrough, we'll process one of its onboard kick sounds to provide the solid 4/4 rhythm that we need.

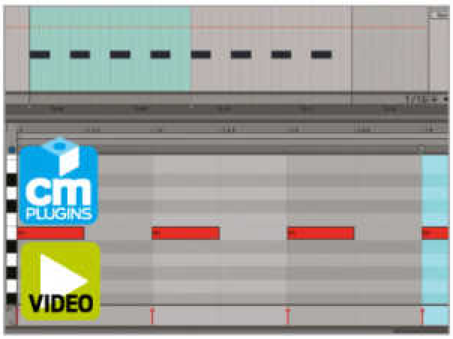
After crafting and processing the percussion parts, we'll synthesise a classic FM bass sound with our own Curve 2 CM, subsequently building upon this foundation with a chord sequence, incidental sounds and a chopped vocal element.

With the track written, we'll apply mix and master processing to the tune, before finally arranging it out in a club-friendly structure.

"After crafting the percussion parts, we'll synthesise a classic FM bass sound"

> Step by step

1. Creating a house kick with AudioRealism ADM CM and processing



1 > The kick drum forms the solid four-to-the-floor foundation of any self-respecting house track, so let's begin by making our own using ADM CM's in-built samples. Launch Live, set the project tempo to **124bpm**, then load the drum machine plugin on a new MIDI track. Create a bar-long MIDI region and program a **B0** note on every beat of the bar, each one an eighth-note long.

2 > Click the **menu switch** at the top right of ADM CM and select **Patch Library»Electro**, switch the **Mode** from Ptn to **Note** (so we can play individual sounds via MIDI rather than triggering patterns), then start playback in Live and you'll hear the plugin play back the kick. On the kick's channel (**BD**), set the **Decay** to **1.5s** and **Tuning** to **B-3** (**-1.44sm**). The kick is too boxy and weak for our purposes, so let's beef it up...

3 > Insert CM-EQUA 87 on the channel. Set the LF band to around **180Hz** at **-11dB** (with a **Q** of **1.00**) to pull out that lower mid-range boxiness, making space for later instruments and highlighting the kick's low-end thump. Turn CM-EQUA 87's **Input** up to **1.20dB** to re-level, so the processing isn't causing a drop in volume. Next, add Wolfram CM and push **Dist** up to **10 o'clock**, adding aggression, then turn the plugin's **Out** to **12 o'clock** to re-level.

4 > The kick lacks top-end crack. Set CM-EQUA 87's HF Frequency band to around **2800Hz**, **Q** to the default **0.70** and **Gain** to **+12dB**, boosting the high-mid of the kick. We still need to add a touch more brightness to the attack, and SKnote's Snap is designed for this – add it first in the plugin chain, then set **Body** to **5** and **Hit** to **20**. CM-EQUA 87's limiter and Wolfram CM's distortion help to keep this transient boost from creating overt peaks.

5 > Finally, add Barricade CM last in the plugin chain. An **Out ceiling** of around **-7.8dB** and **Input gain** of **+2.0dB** chop off the transient spike at the start of the hit. Turn the chain of plugins on and off to compare the noticeable effect the processing is having. The track will fill up with instruments and plugins later, so – as we're happy with the kick – we render a two-bar loop of this sample for convenience and to save CPU.

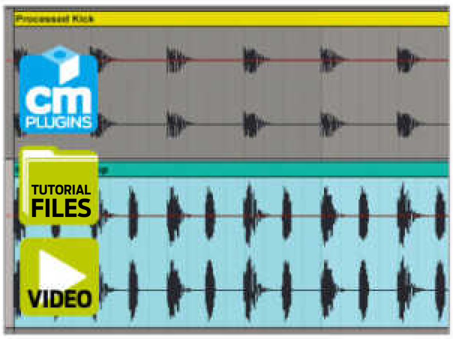
POWER TIP

> Low-end balance

The low frequency relationship between a house kick drum and bassline is one of the most important things to consider, and also the most tricky to mix, thanks to the need for a suitable acoustic environment and accurate monitoring. Traditionally, the kick's fundamental frequency sits in the 40-60Hz region, while the bass's would sit above this in the 80-120Hz range; but trends in bass-heavy genres have made modern house kicks shorter and punchier, leaving room for heavier bass frequencies underneath. It's easier to make decisions at the early stages of sound design and composition, so the kick we've created here is fairly short, solid and somewhat subtle, ready to fit with a powerful bass element.

> Step by step

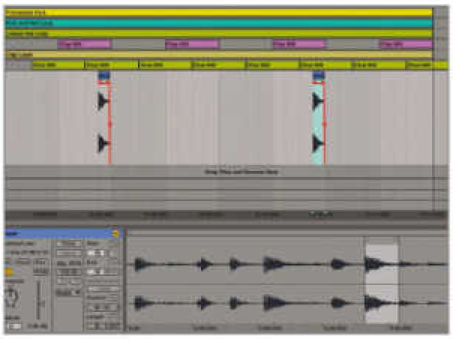
2. Building up a house percussion groove



1 > Now to build a drum groove around the kick drum. Import ours (**Final Kick.wav**) now if you haven't rendered it out yourself from the previous walkthrough. For an initial backing groove, import **Kick and Hat Loop.wav** onto a second audio channel and turn its level down to **-12dB**.

2 > The low frequencies in this loop are clashing with the main kick, so insert DDMF's IIEQ Pro CM on the loop's channel and apply a **HPF 12** – a 12dB per octave high-pass filter – at **465Hz** to resolve this conflict. Now drop **Closed Hat Loop.wav** on a new audio channel, turn it down by **-23dB**, then pan it Left by **10**. Notice how this hi-hat complements the one panned right in our first loop.

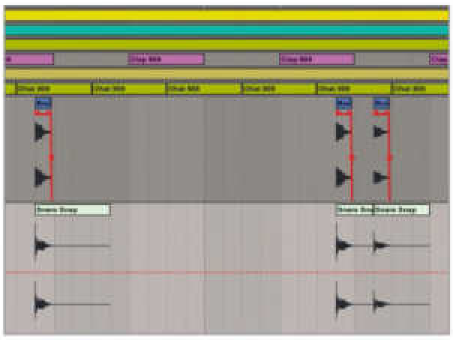
3 > A TR-909 clap sample is a classic house percussion element. Place **Clap 909.wav** on beats 2, 4, 6 and 8 on a new audio track. The sample is clean and quiet, so we load RP-Distort CM on its track and load the **Clap Dist.fxp** preset to dirty things up, then turn the channel down to **-9dB**. Lastly, load Barricade CM, with its **Out ceiling** at **-1.0dB**, squashing the clap for a flatter, limited sound.

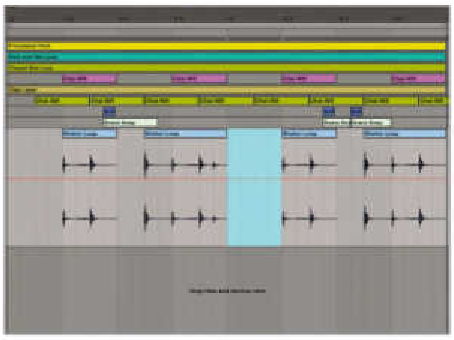


4 > A wide, noisy layer can add character and space. Drop **Clap Layer.wav** on a new channel set to **-17dB**. To remove some clashing low-end, add Satson CM, activate its **12dB/oct** switch and set the **High Pass** filter to **12 o'clock**. Let's limit the layer to blend it behind the first hit – load Barricade CM, set an **Out ceiling** of **-15dB**, **In gain** of **+20dB**, **Attack** at **0.003s** and **Release** at **0.210s**. Pull this layer's track delay back to **-2ms** so it hits just *before* the first clap.

5 > Offbeat hi-hats are another house staple – put **Ohat 909.wav** on the offbeat between each kick. For loudness and grit, set PreMix CM's **Gain** to **+27.5dB** and **Output** to **-2.5dB**. We'll shape the hi-hat's tone by setting **Low** to **9 o'clock** and High to 11 o'clock. The hats sound flat and static, so let's use CM Verb for width and space – set its **Reverb level** to **-2.0dB**, **Medium chamber**, **0.25 time**, **Low cut-off** to **2000Hz** and **High cut-off** to **8000Hz**.

6 > Taking a cue from UK garage, let's add quiet 'ghost' snares to inject funk into the otherwise straight groove. Import **Breakbeat.wav** onto a new channel, turn the level down to **-4.5dB**, switch off quantise, then chop out the main ghost snare (found after the first main snare hit) and delete the rest of the file. Place this hit on the last 16th-note of the first two beats of each bar. Apply a small volume fade to its right edge to prevent clicks.





7 > Copy this snare hit to the first 16th-note of bar 2's beat three, then turn this audio event down by **-4dB**. The hits sound rigid because they're directly on the beat, so set the channel's track **Delay** to **23ms** to push them 'later' in the groove, adding swing. For more crack, load **Snare Snap.wav** and layer it with the snare at **-13dB**.

8 > A subtle shaker rhythm will really drive the track along – import **Shaker Loop.wav** onto a new audio track and turn it down to **-9dB**. This makes the beat too busy, so chop some beats out, as shown. Now, after muting the main kick channel, we render the entire percussion mix as a single audio file for later use. Making your own custom loops in this way can really help kickstart future track ideas.

9 > Now for a wider shaker part to drop in and out as needed. Import **Second Shaker Loop.wav** and level it to **-16dB**. Open RP-Distort CM on its track and load **ShakerWide.fxp**. The free MUtility from MeldaProduction (get it at **meldaproduction.com**) will rein this width in – set **Panorama left** and **Panorama right** to **44%**, raise the **Gain** to **+2.00dB** and render the loop.

Five choice house tracks

Pilo & Sinden, *Night Visions*
This superb release on Boysnoize Records combines chopped-up police sirens, TR-808s, breaks and repeating drum machine beats to achieve a distinctively 80s electro vibe. It's a dirty rave number packed with reverses, chops and edits.
youtu.be/O9oUcMjJJYY

Henry Crinkle, *Stay (Justin Martin Remix)*
A classy remix that takes the original's blissful chords and vocal and places them over a cooler house rhythm. We love how the sparse bass, distorted clap and high-passed breakbeat keep the track rolling along as it evolves through sections of the arrangement.
youtu.be/PoKZlyaDAmQ

Dusky, *Inta*
This cleverly-named track uses the famous "enter" vocal sample, sitting it over retro beats, house piano chords and lush strings. However, it's the huge horn-style bass tone that steals the show here, pushing out plenty of weight to keep club systems working to maximum capacity.
youtu.be/-04NqbadQj8

Sidney Charles, *Hurricane*
German producer Sidney Charles makes modern house music with a nod to the past. Funk stabs, head-nodding beats, a funk-fuelled bass loop and a clever spoken vocal combine to create this cool head-nodder – and the sampled funk stabs provide the icing on the cake.
youtu.be/uPZfwXW9_z4

Mark Knight, *In and Out*
This chugging workout makes use of the classic M1 organ bass patch to full effect, combining it with rolling beats, piano hits, saxophone stabs and a recognisable female vocal hook. Everything combines to make it an effective looping track for the dancefloor.
youtu.be/lla7EP-pGwM

> Step by step

3. Synthesising a deep house-style FM bass

1 > It's time to create an FM (frequency modulation) synthesis bass tone that complements our 90s-inspired crunchy drums. Open a blank 124bpm project and import the rendered kick and percussion parts previously created onto separate audio channels, grouping/routing them to a single 'Beats' group channel. Now create a new MIDI track and load a fresh instance of Cableguys' Curve 2 CM on it. Click the splash screen to access the main interface.

2 > Our short, tight and punchy kick leaves us plenty of room for a weighty bass. Low E to G notes give the maximum low-end impact for the 40-60Hz region, so we trigger a low E1 while programming the patch. Click **New** at the top to initialise the synth, then raise the bottom **Preset Volume** value up to a maximum **6.00dB**. Set **Unisono** to **1** and **Play Mode** to **Mono** – this gives us a standard, single-voice, monophonic patch.

3 > Let's use Curve 2 CM's FM capabilities to craft a bass tone. Turn Osc 1's **Volume** down to **0.00** – we'll use it to modulate the audible Osc 2 sine wave 'behind the scenes'. Pull Osc 2 (Wave 3)'s **Volume** up to **0.60** and tune its **Pitch** down to **-12** for a lower sine tone. In the OSC2 FM section, increase the **EG*OSC1** parameter – note how the first oscillator modulates the second by an amount defined by the envelope generator (EG).

4 > Leave **EG*OSC1** at **36**. The current settings are creating a warped garage-style bass, so let's tweak the envelope's shape to create a pluckier 'donk bass' effect. Tune Osc 1's **Pitch** up to **+12**, then change the EG's settings: set **Attack** to **0.00ms**, **Decay** to **545ms**, **Loop** to **Off** and **Release** to **211ms**. It's still not snappy enough, so click the EG's **spyglass** icon to enter the waveform editor.

5 > Click the graph display to create a node, and set it to **x: 2750**, **y: 0.00**. Then create another point and set it to **x: 500**, **y: 0.00**. Tweak these settings to create a tight, percussive FM pluck bass. To complete the patch, set the volume envelope (Vol-EG) as follows: **Attack 0.30ms**, **Decay 885ms**, **Loop 32.00s**, **Release 565ms**.

POWER TIP

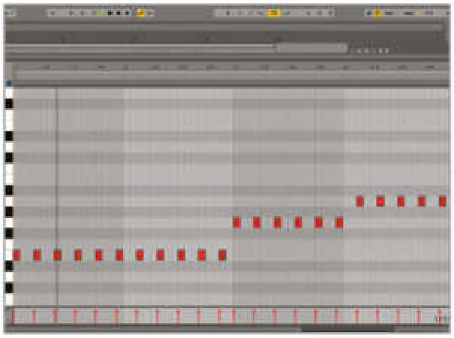
> Bass layering

Often, a single sound source may fulfil certain sonic characteristics but lack others. By stacking two or more signals together, we can combine their characteristics into one full-frequency, pro-sounding element. Our Curve 2 CM bass hits the right low frequencies but needs a touch of wide, trebly attack. Instead of fiddling with the patch, it's easier to create a high-passed 'pluck' layer to add this top-end detail. A third, wide (and, again, high-passed) layer can also be brought in and out later to change the bass's tone, for a refreshing change when necessary.

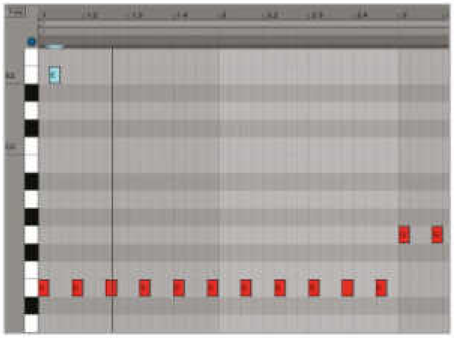
> Step by step

4. House bass programming, layering and processing

1 > We've synthesised our bass from scratch, so let's do the tone justice by programming a memorable riff to go with the beat. Create a four-bar MIDI region on the bass synth's track. Inside the MIDI region, apply a groove template – **MPC 16 Swing-62** – to delay every second 16th-note slightly for a swung feel. This groove works well with the swing of our drums. Most DAWs feature some kind of swing function similar to this.



2 > As we alluded to earlier, the low E1 note (approximately 41Hz) is about as far down the keyboard as you can play before a bass tone becomes too deep for a sound system to accurately reproduce and ears to clearly discern, so let's use E as the root key (or tonic) for maximum weight. We start our riff with the lowest **E1** for two bars, then move the third bar's notes up to **G1** and the fourth bar's notes up to **A1** – each one a 16th-note in length.



3 > The rhythm is based on repetitive groupings of three 16th-notes – a classic house trick. We add an extra high E2 note on the second 16th-note of the riff's first beat for interest. Our final riff is basic but effective, interplaying with the beats both rhythmically and melodically. The simple rising progression also gives us scope to add chords and stab progressions later. The final MIDI file is in the **Tutorial Files** folder – **Bass Riff.mid**.

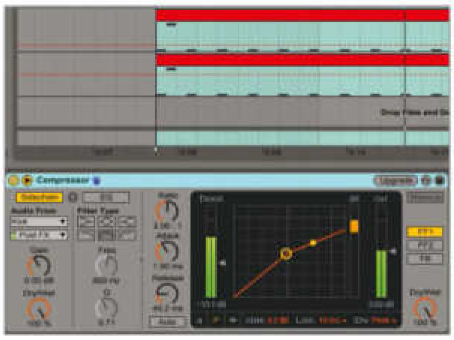
4 > Our core bass is a simple mono patch, so let's add a subtle pluck layer for character and stereo space. Load a fresh instance of Synapse Audio Dune CM onto a new MIDI track, duplicate the bass's MIDI region onto it, then load the **025: Electro Kitty RL** preset. Switch both of the synth's oscillators to a **Saw wave**, pull the Amp Envelope's **Decay** down to **44%** for more 'pluck', then pull **Glide** down to a minimum **0%** to disable portamento.

5 > Load Satson CM on the new layer's channel, flip the **12dB/oct** switch, then set **High Pass** to **1.8kHz** and **Low Pass** to **1kHz**. This band-passes the layer, stopping it clashing with the main FM bass. Next, on the same channel, we load Barricade CM, setting **In gain** to **+10.4dB** and **Out ceiling** to **-16.5dB** to prevent the signal's dynamics from jumping around. This layer adds subtle treble detail and width to our main bass.

6 > We now load an instance of LinPlug's Alpha CM onto a new MIDI track, then copy the MIDI file of our bass riff over to this track and load Alpha CM's **Bass Layer.fxp** preset file (found in the **Tutorial Files** folder). An instance of Satson CM – with a **12dB/oct High Pass** set to 1 o'clock and **Low Pass** set to **1kHz** – fits this wide, warping layer in with our other elements.

7 > Now let's process our main Curve 2 CM bass. Load CM-EQUA 87 onto its track and turn off the **Lo Cut** filter so that it's not removing any low end. Now set the LF's **Frequency** to **80Hz**, **Q** to **0.70** and **Gain** to **-2.8dB**, gently removing low end. The MF band is then set to **300Hz**, its Q is set to **2.90**, and a Gain of **-5.2dB** pulls out those honky lower mids.

8 > Insert two instances of Satson CM: the first with **Gain** set to 1-2 o'clock and the second at 9-10 o'clock. In between these plugins, load a CM-COMP 87 and push its **Level** to **6.00dB**. By doing this, we're pushing the signal into the compressor's output limiter, levelling the dynamics. Now group the three bass channels to a single group channel named 'Bass'.

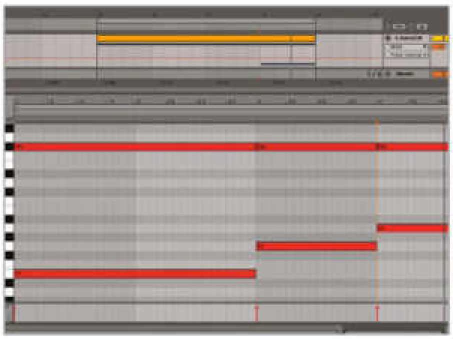


9 > To duck the bass whenever the kick fires, strap Live 9's **Compressor** across the Bass group and route the kick to its sidechain input. A **Threshold** of **-19dB**, **2:1 Ratio**, **1ms Attack** and **50ms Release** only provide a touch of ducking, as this is meant to be more of a mix aid for reducing clashes between the kick and bass than an obvious pumping effect.

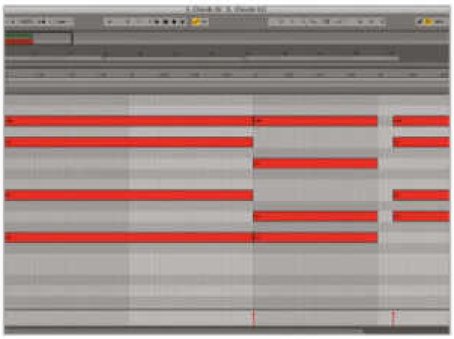
> Step by step 5. Crafting a lush house chord progression

1 > We have a solid beat and bass groove now, so let's create a main chord progression over the top – this will form the musical backbone of our track. Duplicate all the parts to make a 16-bar loop. This will make life easier, since we'll be arranging the song in 16-bar sections.

2 > Add u-he's Zebra CM on a new MIDI track, turn the track down to **-12dB**, then load up the **CM Pad.fxp** preset from the **Tutorial Files** folder – a lush pad patch we've customised for this track. Instructions for loading the patch can be found in a text file along with the preset in **Tutorial Files**. Create a new four-bar MIDI region on this channel and loop it up.



3 > To follow the musical progression of the E-G-A bass notes (created earlier), we draw in a two-bar **E3** note first, then a one-bar note on **G3** and a final one-bar note on **A3**. For a slightly melancholic feel, we duplicate these notes and put each copy up to **F♯4**.



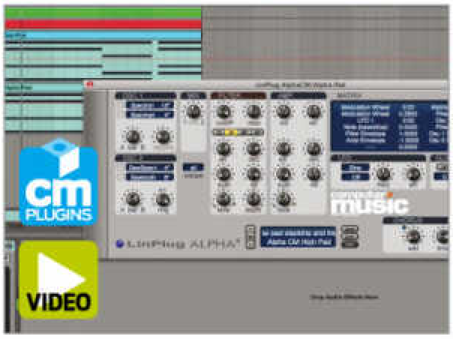
4 > We then add two extra notes to each of the three chords, as shown, making the final progression Em(add9), Gmaj7sus2 (essentially a D chord with a G in the bass) and A sus2 add6 – phew! We've put all the notes in the same octave. Drag the notes of the final chord later by an eighth-note to give more rhythmic interest to the riff.

Nice pad

A pad is really just a series of long, sustained synth notes used to provide interesting timbral movement and width in a musical piece. You need two things to create great pads: a suitable combination of notes (ie, a chord progression, be it a simple or complex one) and an interesting synth tone. Luckily, our free cm Plugins collection is packed with awesome pad-creating synthesiser sources.

In this feature, we've enlisted Zebra CM and Alpha CM to provide us with intriguing, floaty pad sounds, but you can coax these kinds of tones out of most other synths from cm Plugins. XILS-lab's PolyKB II CM is a brilliant choice for the more traditional analogue-style pad or string sound we're used to hearing from vintage analogue synths by Moog, ARP, Sequential Circuits and the like. Enzyme CM – Humanoid Sound Systems' scanned synthesis instrument – is perfect for colder, evolving, digital pad parts. Camel Audio's Alchemy CM houses heaps of inspiring pad and string presets, and Synapse Audio's Dune CM is ideal for smooth, subtractive pads, though you'll have to add external effects to the raw tone.

> Step by step 6. House pad stacking and treatment



1 > A second, higher pad element on top of the first one will add further high-end texture, so load Alpha CM onto a new MIDI track, turn it down to **-14dB**, then copy the MIDI region from the Zebra CM track and transpose all the notes up an octave. In the **Tutorial Files** folder, you'll find **Alpha CM High Pad.fxp** – load this patch into the synth via the little 'folder' button at the bottom-left of the plugin.

2 > This new pad sound is mono, centred and not terribly exciting, so we're going to widen it out and add character using further processing. Insert Wolfram CM onto the channel, then load the **Chorus – CM Fat Bastard** factory preset. Pull the **Mix** control back to around **2 o'clock** and add a touch of **Out gain** for a smoother blend of wet and dry signals.

3 > To remove undesirable low end from the pads and prevent them interfering with the bass, group them in a single 'Pads' channel and insert Satson CM with the **High Pass** dial set to around **11 o'clock**. Now to get them pumping with sidechain compression! With Live's Compressor on the group, route the kick to its sidechain input, and set the **Threshold** to **-33dB** and **Release** to **10.2ms**.

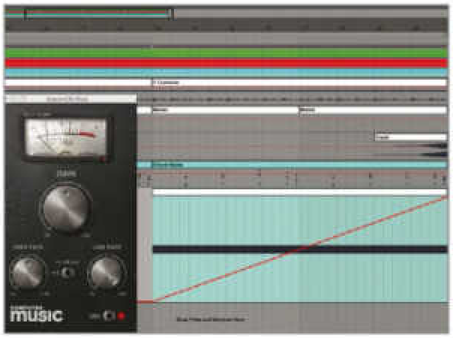
> Step by step

7. Filling out a house groove with stabs, chords and FX

1 > Time to embellish our groove with cool incidental sounds. Load Loomer's Cumulus on a new MIDI track, call up the **Aqua Dream** factory preset and turn the channel fader down to **-9.5dB**. We record eight bars of this rhythmic FX loop as audio, deleting the synth itself when we're done. A Satson CM on this new audio track, with its **High Pass** set to **1.8kHz**, rolls off the bass and mids to put the sound in its own space within the mix.

2 > Now fire up Camel Audio's Alchemy Player CM on a new MIDI track, then load the **Factory»Mallets»Malletarium A** preset. Create a four-bar MIDI region on this track containing an E2 note on the offbeat after the seventh beat (note 2.3.3). This characterful delayed stab sits between the other track elements nicely. Push Alchemy Player CM's **7 - Stereo** parameter up to **40%** for width.

3 > Let's create a reverse cymbal to help move between sections. Load **Crash.wav** onto a new track and turn its channel down to **-11.5dB**. Reverse the audio file, then place it at the end of the 16-bar section. Insert Eurydice CM onto the track, load the **CM_DelayOnly** preset, then set the **Delay Time** to about **170ms** and **Feedback** to **85%** for some floaty echoes.



4 > Now for some piano-ish chordal stabs. Load Alchemy CM onto a new MIDI track, call up the **Factory Keys Dulcitone Layered** patch and drop **Chord Stabs.mid** in from **Tutorial Files**. For a bigger sound, turn **Body** up to **100%**, **Cutoff** down to **4.5%**, **Res** up to **100%**, **Rev Mix** to **50%** and **Rev Time** to **80%**. That helps the chords stand out behind the pads.

5 > Insert Satson CM onto the track, then set the (12dB/octave) **High Pass** to 3 o'clock and **Low Pass** to 9 o'clock to roll off the lows and highs. These piano-esque chords sound relatively dry, so insert a Eurydice CM onto the channel, load the **CM_DelayOnly** factory preset, and set the **Delay Time** to **185ms** and **Feedback** to **75%** to make them ring out. Push the channel's level up to **+2.5dB**.

6 > Finally, let's make a pitched riser with Enzyme CM. Load the synth onto a new MIDI track, create an empty eight-bar MIDI clip and draw an eight-bar-long E3 note. Load the **ATMOS_Metal Core** preset, then automate the **Performance 5»Pitch** parameter from minimum to maximum over the eight bars for a zapping riser effect. Satson CM, with its **High Pass** set to 12 o'clock, removes any rumbling lows.

Incidental sounds

House music generally revolves around a solid beat and bass groove, but yours won't sound all that impressive if your other elements are weak or non-existent. Extra stabs, textures, FX and melodic phrases can make the difference between a poor track and a pro one, with even the most minimal house tracks featuring subtle spacey elements, sweeps or crashes.

At this stage of track-building, it's handy to have a custom library of sounds to draw upon - either designed in a previous sound design session, or unused parts rendered out of old projects. That's a whole article in itself, though, so here we've drawn on a handful of our most inspirational cm Plugins to help us instantly source characterful extra sounds that fit in with our bass and drums.

Alchemy Player CM is a powerful synth/sampler instrument pre-loaded with a heap of inspiring sounds in all categories, making it perfect for sourcing incidental effects and stabs - you can get a sense of its versatility above as we modify a piano sound and mallet stab.

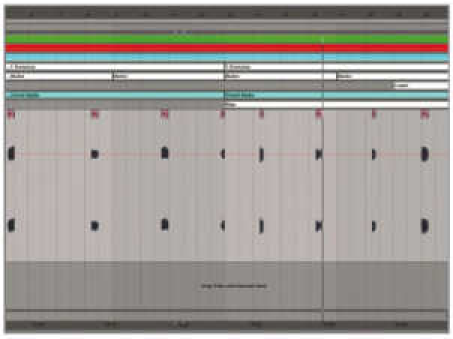
Loomer's Cumulus, an instrument exclusive to cm, is another inspiring source of sounds. It's a granular sampler, so you can import audio files for chopping up then play with the resulting sonic 'particles'. As well as being able to work with your own samples, Cumulus also includes a collection of presets that can be fired up, modified and added to a composition. Give it a try and see if you can come up with something quickly that'll fit.

Delay and reverb are also important tools when creating less obvious FX and stabs, as they help to sit them back in the mix, adding a sense of space and depth. Acon Digital's CM Verb, LiquidSonics' Reverberate CM, Inear Display's Eurydice CM and Subsonic Labs' Wolfram CM are all ideal candidates for this type of spatial processing.

You'll notice that we've applied liberal high-pass filtering to all of these parts using long-time cm Plugins favourite Satson CM to make sure that stabs, cymbals and FX layers don't interfere with the core track elements' (beats, bass and chords) all-important bass and lower-mid frequencies.

> Step by step

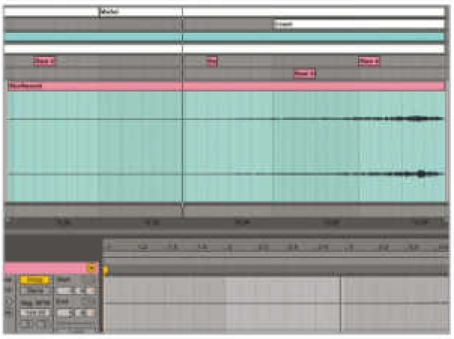
8. Processed house-style vocals



1 > Now let's create a chopped-up vocal element. Drop **Raw Vocal.wav** onto a new audio track, then route it to a new 'Vocal' group channel. In the **Tutorial Files** folder are two Wolfram CM presets: **1 Vocal Pitched.sub** and **2 Vocal Delay.sub**. Follow the instructions in the text file to load the presets into the plugin's library.

2 > Insert two new instances of Wolfram CM on the vocal group's channel, then open the first preset (**1 Vocal Pitched.sub**) in the first one and the second preset (**2 Vocal Delay.sub**) in the second one. These effects blend a pitched-down layer and a series of delay taps in with the dry vocal.

3 > Experiment with a few different chops and small sections of vocal, such as "ooh", "yeah" and other chunks extracted from the full phrase. We can shorten or lengthen these, and the quarter-note delay effect creates rhythmic interplay between them and the groove.



4 > Let's create a floaty, reverbed vocal FX hit. Create a new audio track, duplicate the vocal to it, chop out a small "ooh" section and delete the rest. Now add CM Verb, deactivate its **Dry level** section and crank the **Reverb Time** up to around **12s**. Mute this channel for now, and make sure it isn't routed to the processed Vocal group (so that it isn't processed by the Wolfram CM delays).

5 > For variation, record a portion of this reverbed vocal signal onto a fresh audio track. Reverse the new audio file, creating the classic 'sucking' reverse reverb effect. Position this so that it builds up to the end of a 16-bar section. We'll mute this effect for now, saving it for use later in our arrangement.

POWER TIP

> Multi-effective

When you're knee-deep in creating a track, a workhorse multi-effects plugin is invaluable. You don't have to root through your list of plugins to find separate tools for each job; instead, you can fire up a single device that caters for multiple needs, keeping the creative juices flowing. Here, we've called up one of the most flexible and versatile tools in the cm Plugins arsenal: Wolfram CM. A multitalented wonder, this one offers pitchshifting, distortion, panning, phase shift, delay, filtering, modulation effects and more. It's ideal for vocal processing, and its presets allow for fast mixing decisions.

Sourcing and processing house vocals

Chopping and processing sampled vocal elements has been a staple technique in house music since the genre's inception, and remains as ubiquitous as ever today thanks to the popularity of house acts like MK, Shadow Child, Jamie Jones and Dusky. These artists all make use of modern sampling and audio editing technology to sprinkle retro-inspired vocal cuts, cool chops and eclectic effects throughout their productions.

Obviously, you need to start with a vocal source, and a popular solution is the a cappella (unaccompanied, 'solo' vocal), and – wouldn't you know it? – there are loads of them available online. There are two advantages to using an a cappella: first, if you're chopping the vocal up into small segments, then the credibility of the original source won't matter as much – you can pillage the archives of cheesy vocals for short vowels, licks and phrases. Secondly, your a cappella will probably already feature a 'sampled' tone, automatically giving it that 90s-inspired edge. Copyright could be an issue, however, so you might need to go down a different route. Try recording your own vocalist and applying your own 'sampled' flavour to the audio to crust up the clean recording, turning it into a faux-'ripped' a cappella vocal track

Once you have a vocal source, it's time to chop it up and process it effectively. Nowadays, it's easier than ever to isolate small segments of a vocal phrase – like we've done in the above walkthrough – and piece them back together into a simple rhythmic and/or melodic pattern that can sit happily between the other track elements.

We've edited the audio in Live to chop up our vocal, but you can always turn to a sampler – such as our own XFadeLooper CM – or a dedicated audio editing application to achieve a similar result.

Finally, apply liberal amounts of creative pitchshifting, layering, spatial effects and processing – using Live's own plugins and third party alternatives – to take the dry edge off the vocal. This may give the sound a better 'fit' in the mix and can be a great way to hit upon sonic inspiration.

> Step by step

9. Adjusting the mix of a house track

1 > Before we arrange our track, let's take time out to get the mix in order – this will help us stay inspired while arranging. We'll start with the grouped beats. Insert Satson CM on the group and pull the **Low Pass** knob down slightly (to the rightmost notch) to smoothly roll off harsh treble. Barricade CM – with an **Out ceiling** of **-7dB** – then limits the drums with about 2.5dB of gain reduction when the clap hits.

2 > Next, let's blend in some parallel saturation. To do this, insert PreMix CM on a new return track, turn the channel fader down to **-6dB** and send the Drums group to the return by the maximum amount: **OdB**. Push PreMix CM's **Gain** up to around **+26dB**, then back the **Output** off to **-15dB**. This parallel saturation ups the 'average weight' of the beats for more perceived volume and crunch.

3 > It's apparent at this stage that the kick drum could sit better with the bass, so let's make some EQ adjustments. Insert IIEQ Pro CM on the kick channel and apply a **HPF12** at **22Hz** and a broad **-1.1dB** dip at **122Hz** (at **0.50 Q**). The kick has lost a little low-end impact, but this allows the heavy bass to dominate the low end of the mix, as it should. Now let's widen the percussion loop using the Haas effect…

4 > Create a new return, add Wolfram CM and send the percussion to it. Turn **Manipulator 1 off** and the **Filter/Delay on**. Set delay **Mix** to max and **Feedback** to minimum, pull **Delay Time** down to **50ms**, and turn Offset (**OS**) up to slightly space out the delay times of the left and right channels. The max delay time of 50ms is too long for a pseudo-stereo effect, so set the return track's delay to **-38ms** for an overall delay of just 12ms.

5 > Satson CM, with its **High Pass** set to 1-2 o'clock, attenuates the lows in the widened percussion. To rein in the stereo width on the piano chords and Zebra CM pad, insert MeldaProduction's MUtility on both tracks. For the piano, pull **Panorama left** and **right** in to roughly **45%**, then add **5dB** of makeup **Gain**. For the pad, set **Panorama left** and **right** to around **55%** with about **0.3dB** of **Gain**.

6 > Now let's widen the main bass. Create another return and insert Reverberate CM onto it, then pull the **IR Gain** back to **-35dB** and set the **Dry/Wet** mix fully **Wet**. Load the **Factory»Smallroom Blumlein02** impulse response, and send the Bass group to the reverb at **OdB**. Satson CM's **High Pass** set to **1 o'clock** prevents the return channel's signal from clashing with the bass in the mix.

Three cm mixing tools

VENGEANCE SOUND PHILTA CM

This popular cm Plugin is an all-round frequency-sculpting tool, offering dual resonant high- and low-pass filters, each switchable between 12, 24, 48 or 96dB per octave modes. With variable resonance, width, notch mode and frequency analyser display, it's as at home tightly sculpting a mix as it is creatively sweeping through tracks and FX.

HORNET FAT-FET

With an algorithm modelled on the classic Urei 1176LN Compressor/ Limiter, this analogue-style compressor plugin packs a serious punch. Unlike the 1176LN hardware, Fat-FET features variable threshold and ratio parameters, as well as an ultra-short attack time that reaches down to just 0.2ms – perfect for levelling over-enthusiastic signals.

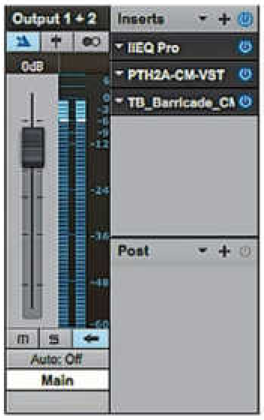
PHOTOSOUNDER SPIRAL CM

It can be difficult to determine the exact harmonic content contained within a sound. Spiral CM tackles this by plotting an incoming signal's harmonics across its colourful circular graph display, enabling you to discern the notes contained within a chord, giving you insight into harmonic relationships and helping you determine the notes in chords.

Method in the mastering

We've approached the mastering process in a slightly unorthodox manner in this tutorial. Generally, you'll create, arrange and finish your track before applying any master processing; but here we've done things the other way around, adding a touch of master buss processing before we create the final arrangement. This workflow approach has a couple of advantages, the first being that our tune already sounds more in line with other commercial releases, so we can reference the arrangement against other mastered songs without turning them down (or our project up). Our track's mastered 'sheen' also provides inspiration as we lay out its various sections – we're all used to listening to mastered material every day, so by mastering your track prior to arranging, it's less likely you'll get disheartened by tonal or level discrepancies, keeping a 'pro' vibe going throughout the arrangement stage. We can always adjust or reapply master buss settings later, or remove any applied master processing and render an unmastered file once the arrangement is done, all ready to send to a professional mastering engineer if necessary.

Of course, we're not suggesting that our mastering method is foolproof. Creating a track is an organic and personal journey, and every project requires a bespoke approach at each stage of its creation – what works for one producer or track might not work for another. We suggest giving our method a go, but you might achieve better results by giving your ears more time to adjust to the arrangement before adding mastering polish. Be sure to experiment and find your own preferred way of working.



Remember, you can always export your stereo mix and master it in a fresh, empty project

> Step by step

10. Final master bus processing a house track

1 > We've processed and mixed the track while creating it, so we only need a couple of final touches to match it up tonally and level-wise to commercial releases. First, insert IIEQ Pro CM on the master channel. A **-1.0dB High Shelf** at **4646Hz** dips the high-mid and treble frequencies a touch, while a **-2.0dB AnaPeak** dip at **255Hz** pulls out some mid-range, cleaning the mix up a bit.

2 > A Pultec-style EQ is often a good choice for gentle-but-characterful frequency adjustment at the mastering stage, so we load OverTone DSP's Program EQ CM next in the chain. Turning the plugin **On**, we set a **CPS** frequency of **100Hz**. A **Boost** of **2.0** and **1.0 Attenuation** apply a Pultec-style combined low boost and lower-mid dip for subtle weight adjustment.

3 > Set the **KCS** frequency to **5kHz**, then apply a **Boost** of **2.5** in that area. By raising the High **Atten**uate knob to **1.0**, we apply a high-shelf cut from around 10kHz. This enhances the upper-mid frequencies while shelving down unnecessary treble from the overall mix, tonally tilting the mix towards the upper mids without allowing the top-end to get too carried away.

4 > A brickwall limiter is generally used to raise the average level of a mix, and for this purpose we load ToneBoosters' Barricade CM last in the chain. With its **Input gain** set to **+6.0dB**, we achieve around 3-4dB of gain reduction for the overall mix, increasing its perceived loudness and further controlling the track's dynamics.

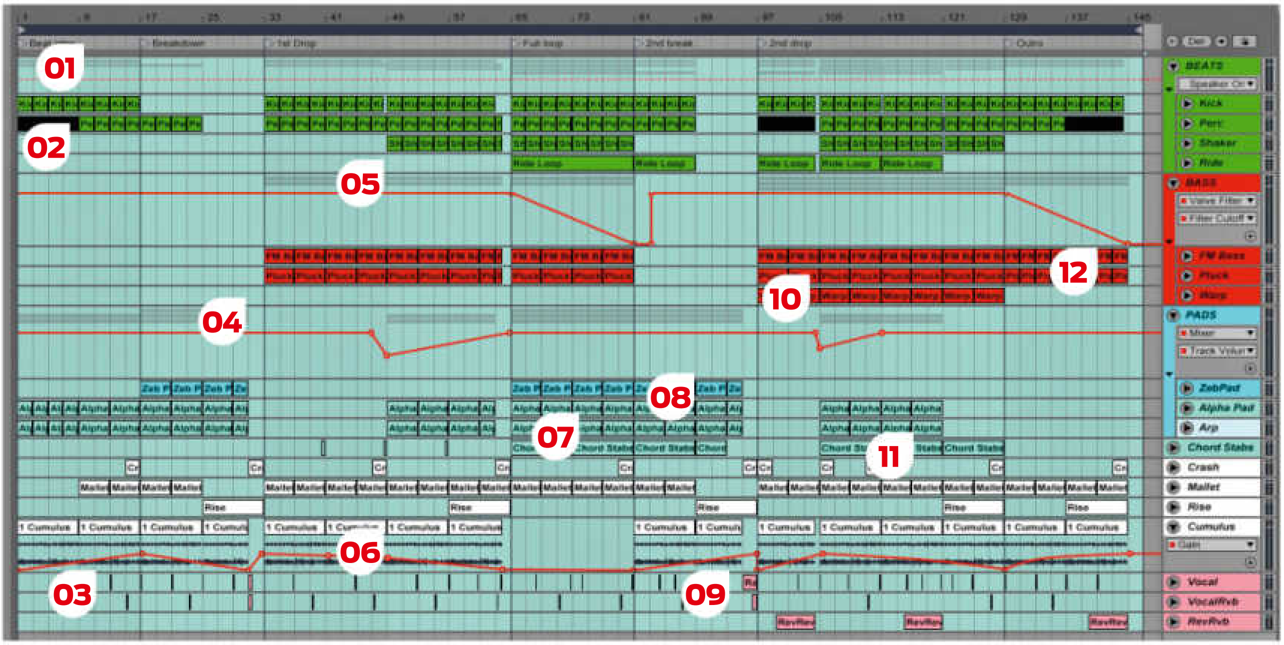
5 > Set the plugin's **Out ceiling** to **-0.5dB** (this helps avoid intersample peaks) before rendering the final stereo master. Our track is complete! Have a cup of tea, put your feet up, and listen through to the final mix for yourself. The mastered and unmastered mixes along with the finished Live 9 project are exclusively available for download in our Vault at **vault.computermusic.co.uk**.

POWER TIP

> Master effects

Master buss processing – that is, mastering – is a widely discussed and somewhat mysterious topic, with the process itself often left to dedicated professionals. It's easy to overcook a carefully-balanced mix with just a couple of overzealous plugin settings, but careful and considered processing can really add that "last 5%" of pro polish to a track. We've mixed the track as we've gone along, so it's already fairly similar in levels to a commercial house track. Therefore each of our master bus plugins only needs apply a very small adjustment – any more than this and the mix may need revisiting.

The final arrangement



01 LOCATORS

To begin, we duplicate the entire 16-bar loop throughout the project, creating Locators at key 16-bar sections and renaming each appropriately. We can then simply delete key elements and repeat sections to create a rough 'subtractive' arrangement.

02 INTRO

We go through the project, arranging sections on the fly. Our 16-bar intro contains just the beat and sparse melodic elements, making life easy for DJs to mix the track. Duplicating short sections of the percussion and pad give a 'repeating' feel.

03 INTRODUCING ELEMENTS

Our vocal chops are introduced gradually, in order to save more elaborate variations for later. The Cumulus FX loop's Gain is faded in, and several parts are brought in – leading the intro into the first breakdown. An arpeggiated riff adds a pro texture over the chords.

04 BREAKDOWN AND BUILD-UP

The kick is removed for the first breakdown, just as we introduce our lower Zebra CM chords. We gradually mute elements throughout the breakdown, until the rise builds and leaves a two-bar gap for the reverse cymbal and a larger chunk of vocal to carry into the drop.

05 FIRST DROP

To give the first drop maximum dancefloor impact, we remove the chord parts here, dropping the track into just beats and bass (and we'll save the stereo warping bass layer for later in the track). We delete drum sections for a basic variation at the end of 16 bars.

06 PROGRESSING THE DROP

Different vocal sections are used over the drop, changing the track's vibe a little. We use small 'tease' sections of the piano stabs, saving the full riff for a more melodic section later. The pad group's volume is faded in via automation, for a more progressive feel.

07 FULL LOOP

Bar 65 is where all of the elements drop in at once. The full piano chords, two pad layers and fresh vocal chops all combine to step the track up. A new ride loop picks up the pace. We use ValveFilter CM to filter the bass, to lead into the second breakdown.

08 SECOND BREAKDOWN

Here, we remove the bass, giving us a sense that the track is cooling down. As in the first breakdown, we gradually remove elements of the track to build anticipation towards the second drop. As our kick is removed here, the pads' sidechain pumping is taken away, opening out the track to let things breathe.

09 SILENCE BEFORE SECOND DROP

Again, we leave a two-bar gap just before things kick back in, but here we use the full vocal – something not yet heard in the track. As with the first drop, we copy a section of the main vocal's audio clip down to the reverb vocal track, for a cavernous vocal effect.

10 SECOND DROP

The stereo bass layer comes in, for a subtle change in timbre. We use the looping percussion trick for eight bars, which, when combined with the ride, gives an offbeat groove. The reverse vocal effect leads us into to a two-beat edit before all hell breaks loose!

11 DEVELOPING THE SECOND DROP

Here we automate our melodic elements back in as we step the track up for the last time. We mute drum elements on the first beat of each eight bar section for rhythmic variation. As with the rest of the track, our piano stabs' tone is evolved via parameter automation.

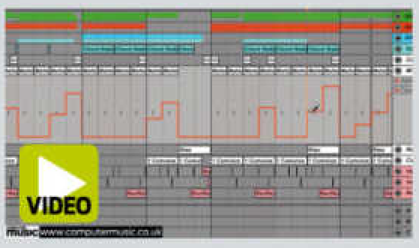
12 OUTRO

To wind the track down, the first two bars of the bass riff are repeated over and over, while being filtered down. The lack of elements plus the repeating percussion make it obvious we're reaching the end of the track, which wraps up with the reverse reverb effect.

Arrangement, polishing up and the final mix

11. CREATING A ROUGH ARRANGEMENT

In this video we copy out the full 16-bar loop, using Live's Locators to mark the beginning of key sections. We move through the track, deleting and repeating various elements to form the bare bones of a club-friendly arrangement.



12. ADDING FINAL POLISH TO OUR HOUSE ARRANGEMENT

After fleshing out a rough structure, we develop a finished arrangement by adding variation and detail throughout. See how we use our chopped-up vocal to full effect, as well as other touches of automation and 'ear candy' that really make the track feel finished.

13. THE FINISHED TRACK

And that's it! Our classic house track is arranged, mixed down, mastered and ready for the club. Download the 24-bit WAV master of our final track and a video playback at **vault.computermusic.co.uk**. You'll also find the finished Ableton Live 9 project to open up and explore at your leisure. **cm**

LIVE SECRETS

Think you know everything there is to know about Ableton Live? Over the next 12 pages, we'll reveal some of its deeper and lesser known functions and capabilities

DOWNLOAD
Get the tutorial files to follow along on your PC/Mac at **vault.computermusic.co.uk**

> If you want to make truly exceptional music with your computer, being able to use your DAW to the best of its abilities is obviously a must. Ableton Live goes out of its way to be accessible and transparent, with a consistent internal logic that makes its advanced operations relatively easy to learn once you've got the hang of the basics. Having said that, it *is* a complicated piece of software, and not everything it can do is immediately apparent. Some functionality can only be accessed by right-clicking device title bars or unfolding a device's parameters – you just have to know where to look.

That's where this guide to Live's hidden features and functions comes in. In the following walkthroughs, we'll show you everything from creative techniques like working with additive synthesis in Live's FM synth Operator and automated pitchshifting effects, to mixing tips including using Live's Vocoder as an enhancer and advanced return track processing. Some of these tips simply use the software's less well-advertised capabilities – even hardcore Live heads might be unaware that it's possible to access hidden options in the software by editing a text file, for example – and others employ its more basic abilities in unusual ways, like using a sidechained Gate to dynamically add white noise to a snare track.

While these tutorials and ideas cover, in our opinion, the best of Live's secrets, there's always something new to learn. Hopefully, we'll give you the inspiration here to think about how Live's flexible architecture can be used to enhance your projects and get you closer to your musical vision. If you're keen to stay on the cutting edge of Ableton's amazing DAW, keep an eye on **www.ableton.com/en/help/article/latest-live-version**, where you can always find out what changes have been made in the most recent version.

9 covert tips & tricks

STEP UP

Live's MIDI Editor can be used as a step sequencer! To do it, arm your MIDI track, create a clip on it, open the clip, ensure the Preview (headphones) button in the MIDI Editor is active (blue), and click anywhere in the grid. Hold down the note or notes you want to play on your QWERTY or MIDI keyboard, and press the right arrow button to add them to the sequence.

QUANTISATION STATION

Auto Filter has a built-in quantisation function that automatically 'steps' the Cutoff parameter. To activate it, click the On button at the bottom of the panel and select a quantisation level using the buttons to the right, each representing the specified number of 16th-notes. Try automating or modulating the Filter Cutoff with quantisation active to create some funky stepped patterns.

Use Auto Filter's built-in quantisation feature to create rhythmic stepped sequences

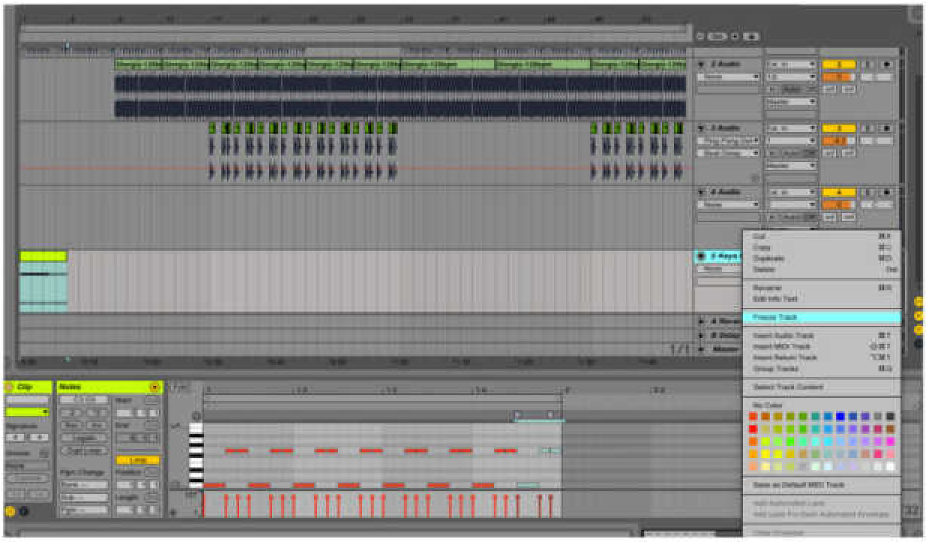
OFF THE GRID

If you're working with snap to grid turned on, the pencil tool will automatically snap automation breakpoints to the grid when you draw them in. This isn't always desirable, but you don't have to toggle snap on and off every time you want to create a smooth automation envelope: simply hold the Alt key while drawing with the pencil tool to prevent it from snapping, enabling freehand drawing.

FREEZE FRAME

Freezing is a convenient way to bounce parts to audio, and if it's something you find yourself doing often, you might want to try dragging the part to be frozen to the beginning of the arrangement before you use the Freeze Track command. As Live renders everything on the track from the beginning of the arrangement to the end of the last clip, this saves on time and disk space wasted through rendering silence before the sound starts playing.

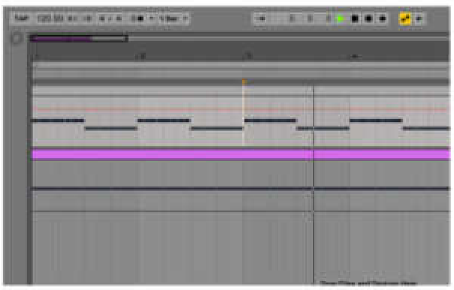
To draw freehand with the pencil tool, rather than snapped to grid, hold down the Alt key



To avoid wasting time and disk space rendering silence, place any parts to be frozen at the start of the arrangement

MAKE A NOTE

Making music can be a complicated business, and sometimes it's tricky to remember the specific things you plan to do with each part of a project. Thankfully, Live has a handy feature for the busy musician in the shape of info text. Info text is essentially a text note that can be added to scenes, clips, tracks, macros, devices, chains and Racks. To add a note to something, select it and choose Edit»Edit Info Text.



Make Live start playing from the current playback point by holding down Shift when you hit the Spacebar

LIGHTEN THE LOAD

If you're working on a heavy project that's ready to export or stem and your computer's CPU is reaching the limits of what it can handle, bear in mind that you can turn Live's audio engine off by clicking the CPU Load Meter. With the audio engine off, Live can't play anything back, but you can still do everything else, including editing and exporting audio, and your system will likely be much more responsive.

MESS WITH YOUR 'HEAD

Live's default behaviour when you click Play or press the Spacebar is to move the playhead to the Arrangement Insert Marker (or the Start Markers of all clips in the Session view) and begin playing from there. To counteract this and start playing from the current playback position, hold down Shift while you press Spacebar.

ALL IN ONE

Big Drum Racks can get confusing when it comes to making adjustments to the devices they contain. If you need to set a parameter to a particular value for multiple instances of an instrument or effect in a Drum Rack at once, simply set it to the correct value on one, then right-click the control and select Copy Value to Siblings.

QUICK CHANGE

Live performers with a taste for the esoteric will be delighted to learn that Live's tempo and time signature can be changed automatically in the Session view by naming Scenes with the relevant information, eg, '3/4, 160BPM' – just the thing for keeping the audience on their toes!

Include a BPM value in the name of a Scene to have Live switch to that tempo when triggered

ReWire selecta!

If you use Live alongside another DAW, you might find its ReWire capabilities helpful. ReWire is a protocol that enables DAWs to transmit audio and MIDI information between each other in real time. So, you could create a MIDI track in Live that triggers a Thor instrument in Propellerhead Reason, the audio output from which is then routed back to an audio track in Live for mixing. Instructions for setting up scenarios like this can be found in Live's manual, but what it won't tell you is that Live has several hidden options that can only be accessed by making and/or editing a particular text file. This allows you to prevent Live from acting as a ReWire master so that another ReWire-enabled DAW can be loaded without the two linking, and specify how many channels appear in the ReWire master when Live is run as a slave.

To get your hands on all this good stuff, create a text file called 'Options.txt' in the folder on your hard drive that contains Live's Preferences.cfg file, and add -ReWireMasterOff or -ReWireChannels=X (where X is 1-64) to the text. Then save the file, launch Live and get to it!

There are plenty of other non-ReWire-related functions that can be accessed this way, too, including the ability to adjust the resolution of recorded automation data and instantly map 'sibling' devices in Drum Racks to macros. See www.ableton.com/en/help/article/optionstxt-file-live for more.

> Step by step 1. Saturator's custom waveshaper

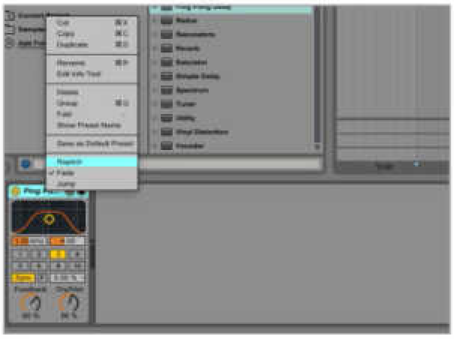
1 > You hopefully already know that Saturator is a great effect with endless uses, but the fact that it includes custom waveshaping controls seems to elude many Live users! Drag **Sine bass.wav** onto an audio track, and add **Saturator** from the **Audio Effects** folder.

2 > Click the **Curve Type** drop-down menu and select **Waveshaper** from the list. On first inspection, Waveshaper appears to be much like the other curves, but click the **Unfold Device Parameters** arrow at the top left-hand corner of the effect's interface to reveal its dirty secret...

3 > Below the curve are six parameters that affect the properties of the waveshaping function. **Depth** superimposes a sine wave onto the curve, the frequency of which is controlled via **Period**. Turn **Depth** up to **100%** and you'll hear that with **Period** at **0.00%**, it merely warms up the sine signal. (Audio: **Warm sine bass.wav**)

4 > However, if you sweep the **Period** control as the signal plays back, you'll get a psychedelic morphing that's reminiscent of frequency modulation and wavetable synthesis! **Drive** controls the influence of the other parameters, so it can be used almost like a low-pass filter to take out the added harmonics. (Audio: **Morphing sine bass.wav**)

> Step by step 2. Dubby tape delay effects



1 > Live's various delay effects (Filter Delay, Ping Pong Delay and Simple Delay) use crossfading when adjusting their delay times to prevent the output from changing in pitch. However, said changes in pitch can actually be used to create awesome, dub-style FX. Drag **Dub siren.wav** onto an audio track.

2 > Now add Ping Pong Delay to the track. Change the effect's transition mode by right-clicking its Device Title Bar and selecting **Repitch**. Now, rather than crossfading, the delay's buffer playback will speed up or slow down when the delay time is changed, much like an old-school tape-based delay.

3 > Change the Delay Mode from **Sync** to **Time**, turn the **Dry/Wet** level down to **50%** and set the **Ms Delay** value to **400ms**. Now play back the **Dub siren** sample and, before the sample finishes, sweep **Ms Delay**. This results in a reggae-tastic twisting effect, ideal for breakdowns and transitions. (Audio: **Dub delay.wav**)

Key Zone splits with Sampler

If there's one thing that Live's not short of, it's ways to cut up, process and otherwise play havoc with audio loops. For the most part, working with loops on audio tracks or having Live slice them into Simpler-triggering MIDI parts automatically (see *Audio slicing and creating custom Library content* on p19 for more on this) will be more than enough to get the job done, so for general production duties, you'll rarely need to get your hands dirty with Sampler. However, the instrument's ability to define Key Zones with a different sample (or part of a sample) assigned to each one, combined with its advanced synthesiser-style processing functions, makes it a powerful tool for creatively processing musical loops.

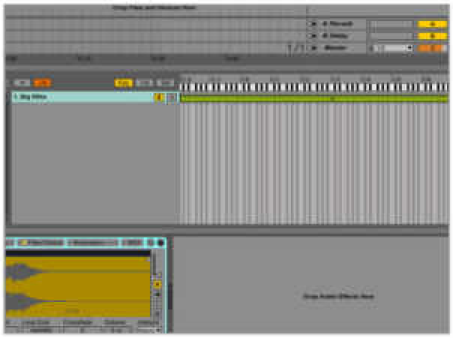
Sampler's Key and Velocity Zones (ie, designated note and velocity ranges for assigning to particular samples) are primarily intended for use in the building of multisampled patches. When making a sampled emulation of a real-world instrument (or, to a lesser extent, a synthesised one), simply transposing a single sampled note across the entire keyboard doesn't work, since a sample repitched just a few semitones away from its natural pitch invariably sounds jarring and unrealistic. However, by using more samples (a separate recording for each note or octave, for example), much more convincing and authentic-sounding results can be achieved – and that, in a nutshell, is multisampling.

If we use Key Zones to assign separate sections of a musical loop to individual keys, we can play it back in new and interesting ways. What's more, we can treat it like an instrument we created ourselves using Sampler's arsenal of sound-sculpting tools, including its amplitude envelope, filter, waveshaper, FM and AM oscillator, and pitch modulation system. All of Sampler's parameters (apart from RootKey) affect all of its Key and Velocity Zones, so you can automate them without having to think about which slice is playing.

In the walkthrough below, we're going to show you how to turn a musical loop into a new sequence that can be morphed and evolved in a synth-like fashion.

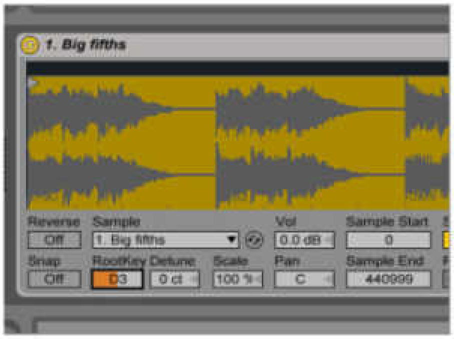
> Step by step

3. Splitting a loop across Key Zones in Sampler

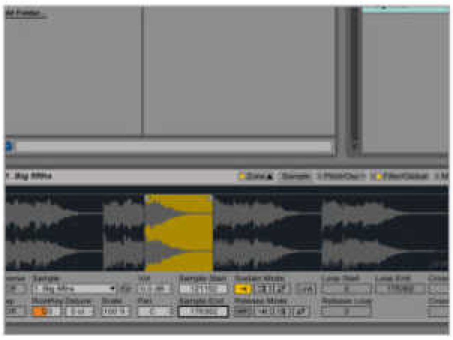


1 > Drag **Sampler** from the **Instruments** folder onto a MIDI track, then drag **Big fifths.wav** onto the **Drop Sample Here** area. Click the **Zone** tab at the top of the interface and Sampler's Zone Editor will appear above the instrument. At the moment, just a single sample is loaded, and we can see from the **Key** range that it plays on every note of the keyboard.

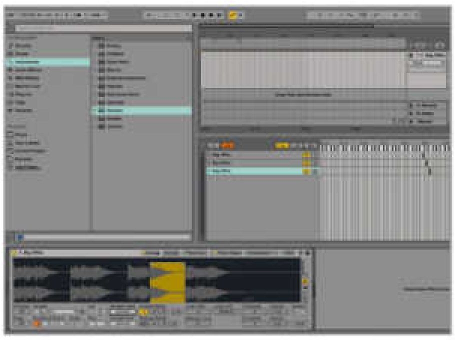
2 > Right-click **Big fifths** in the Sample Layer List and select **Duplicate**. This creates an exact copy of the first layer, again spanning the whole keyboard. We want each layer assigned to its own key, so drag the left- and right-hand edges of the first layer to C3.



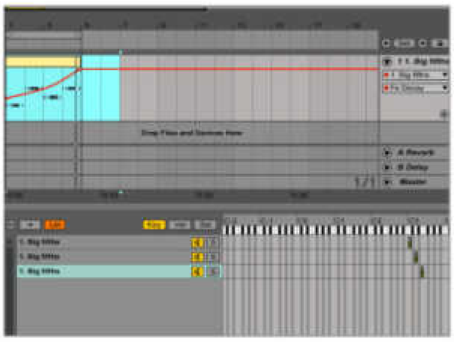
3 > Do the same thing for the second layer, but this time drag the edges to D3 instead. You'll notice that the small R on the track remains on C3. This represents the sample's root note, so when we play note D3, the sample plays back two semitones up from its natural pitch. To correct this, drag the **RootKey** parameter on the Samples page up to **D3**.



4 > Now, both C3 and D3 trigger the same sample at the same pitch. Let's adjust the layers' sample start and end points so that they play different sections of the sample. Click the first layer in the Zone Editor and set the **Sample Start** parameter on the Samples page to **33039** and the **Sample End** to **88180**. Then, click the second layer and set its **Sample Start** to **121192** and **Sample End** to **176362**.



5 > Now the first chord is on C3 and the second is on D3. Duplicate the second layer and set the new version to play on E3, with its root note set to E3, too. Next, set this new layer's **Sample Start** and **Sample End** points to **209463** and **264430** respectively. Now that we've split up all the chords, we can get tweaking.



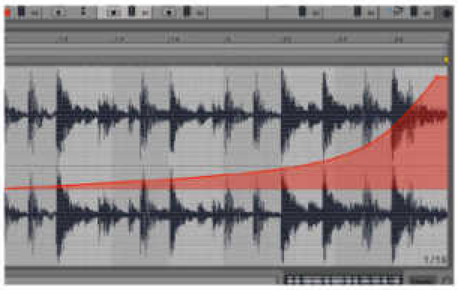
6 > Click the **Filter/Global** tab, and bring the **Filter Freq** down to **200Hz**. Then, set the **F. Env Amount** to **+72** and you'll hear this tight low-pass filter movement affect all the slices we've defined. Program your own riff and try automating the **F. Env Decay** so that the filter opens up more gradually, creating a trancey crescendo. (Audio: **Opening filter.wav**)

Creating FX with warping

Ableton have done a fantastic job in designing Live's warp modes, which deliver amazingly transparent results when applied judiciously. Indeed, so good are its timestretching and pitchshifting that many users of other DAWs turn to Live solely for those processes!

When pushed to extremes, timestretching and pitchshifting can deliver astonishing results. The creation of much of the electronic music of the 90s centred on experimentation with the limits of what could be done using the relatively affordable hardware samplers of the time – in particular, Akai's S series.

The S series' time and pitch processing was simple: stretch a sample offline to a user-defined percentage of its original length, referring to the included table of lengthening or shortening percentages to maintain its original duration



Real-time pitchshifting effects are just one trick that can be applied via Live's sample editor

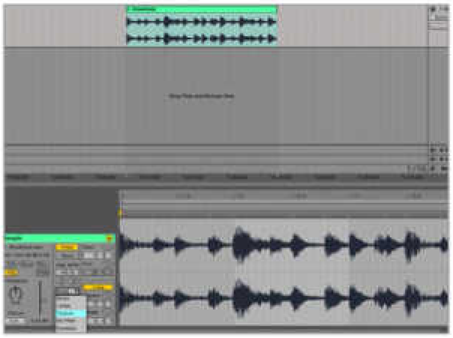
when transposed up or down to a particular pitch. Despite its primitive functionality, the S series' timestretching was used extensively by the producers of the time, and classic tracks such as CJ Bolland's *Sugar Is Sweeter (Armand's Drum And Bass Mix)* and Origin Unknown's *Valley of the Shadows* used prominent vocal and beat timestretching to create totally original abstract effects.

Live's warping is leagues ahead of these now-antiquated machines, of course, but they can still be pushed beyond their apparent limits in the quest for exciting new sounds. It's also possible to emulate old-school Akai-style timestretching with them, although the creative possibilities are far greater thanks to Live's much wider range of tweakable parameters.

In the walkthrough below, we'll show you how to turn beats into psychedelic rhythmic riffs, create turntable brake effects, and contort vocals into hideous new forms – something for everyone, then!

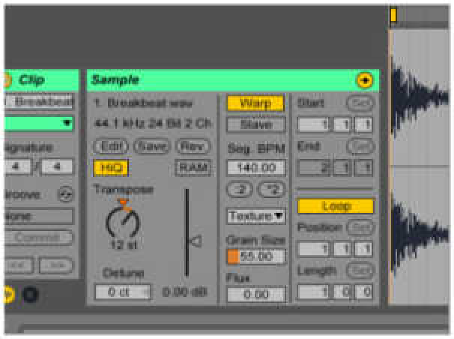
> Step by step

4. Making abstract FX with Live's Warp Modes

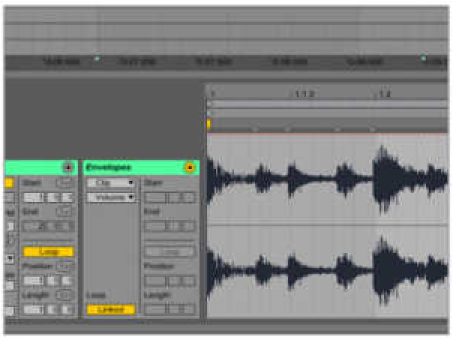


1 > Drag **Breakbeat.wav** onto an audio track and double-click the clip to open the Clip View. Live's default **Warp Mode** is **Beats**, and this works fairly transparently for moderate tempo and pitch changes to rhythmic material. However, we're not interested in transparency, so change the **Warp Mode** to **Texture**, which is intended for atmospheric sounds.

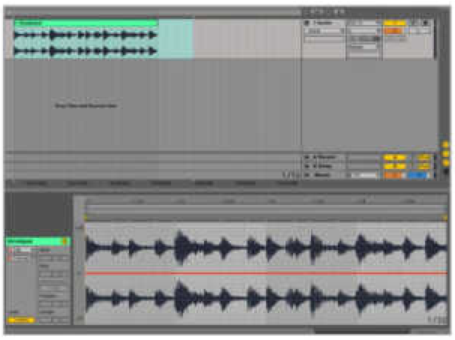
2 > At the loop's original tempo, changing the **Warp Mode** doesn't have a huge effect – until we change the clip's pitch, that is. Set **Transpose** to **+12st** and the break takes on a mysterious, pitched, metallic quality. The **Flux** fader introduces random processing, but a more consistent, uniform sound works better for this effect, so turn it down to **0.00**.



3 > Turn the **Grain Size** down to **2.00**. This creates a strong, all-pass filter-style effect, and if we gradually turn it down further as we play the beat back, we can hear the effect transform into a comb-filter-style sound before arriving at the classic Akai S-series pitchshifting feel. This is optimal at about **55.00**, so set the **Grain Size** to that. (Audio: **Pitched break.wav**)

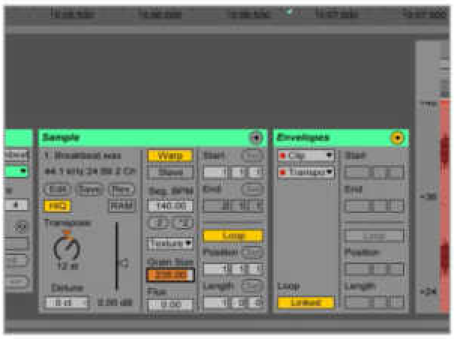


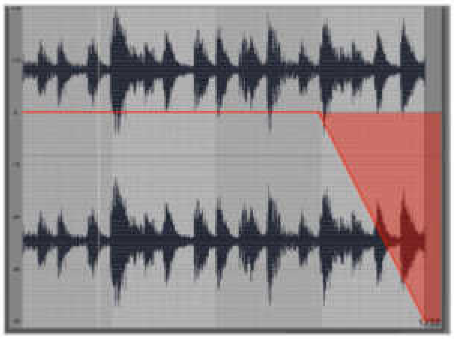
4 > Live is a lot more powerful than an Akai sampler, so we can use it to perform some really cool tricks. For example, Live has no problem pitchshifting audio in real time. Click the **Show/Hide Envelope Box** (**E**) button at the bottom left of the Clip View to bring up the clip's breakpoint envelope.



5 > In the **Envelopes** panel, click the **Control Chooser** drop-down and select **Transposition Modulation**. Here we've set the **Transposition** parameter to **+12**, but this is independent from the Transposition Modulation envelope, which you'll see sits at **0** for the duration of the clip. To add a breakpoint, click the envelope. Add two breakpoints now.

6 > Move the breakpoints so that the Transposition Modulation rises from **0** to **12** over the last half of the clip. Play the project back and you'll hear the pitch of the beat rise with the envelope, creating a psychedelic effect. (Audio: **Rising pitch.wav**, **Extreme pitchshifting.wav**)

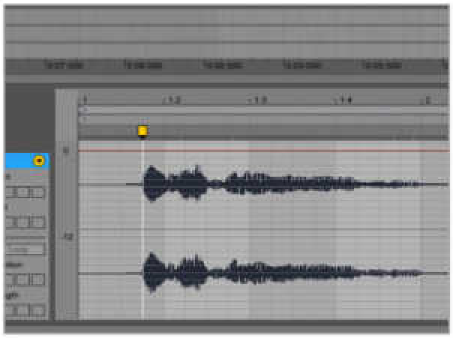


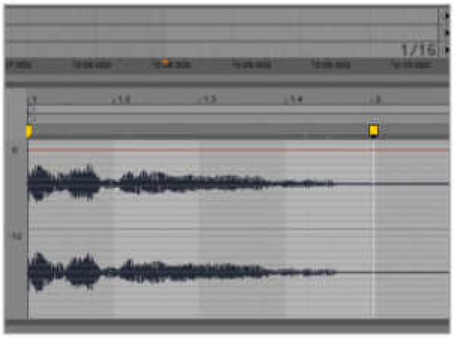


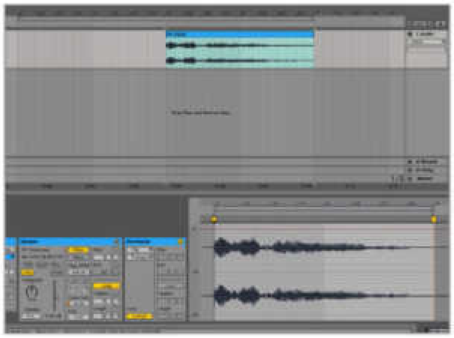
7 > Using more extreme settings can produce a vocoder-like effect. Set the envelope so that the Transposition Modulation sits at **+48 semitones** for the entire length of the clip. With this kind of dramatic transposition, even small changes to the **Grain Size** setting will result in wildly different sounds – turn it down to **50.00** to hear this for yourself. (Audio: **Extreme pitchshifting.wav**)

8 > Turning the **Grain Size** up causes the sound to lose all sense of rhythm and collapse into a glitchy texture. Set the **Grain Size** to **235.00** to check this out. Another, more traditional effect that can be created using transposition modulation is the Logic-style 'speed fade'. (Audio: **Glitchy texture.wav**)

9 > Right-click the Sample Editor and select **Clear Envelope**, then return the **Transpose** knob to **0 semitones**, so that the beat plays back normally. Change the **Warp Mode** to **Complex** and create a drop in pitch from **0** to **-48** semitones over the last beat, as shown above. This emulates the sound of a turntable braking. (Audio: **Turntable brake.wav**)



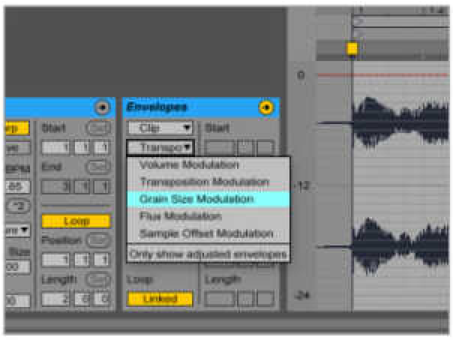


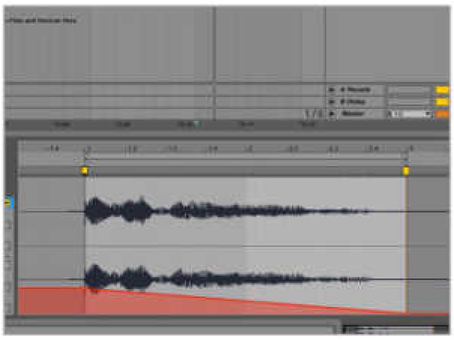


10 > Another cool warping effect we can try is the old-school jungle/garage-style stretched vocal. Delete **Breakbeat.wav** and drag **Vocal.wav** onto the audio track. In the Sample Editor, double-click the existing Warp Marker to delete it, then double-click the transient at the start of the sample to create a Warp Marker there.

11 > Drag the Warp Marker to the start of bar 1, then press **Ctrl/Cmd+4** to deactivate Live's snap to grid mode. Delete the Warp Marker at the end of the sample, then double-click the Transient strip at the start of bar 2 to create a new Warp Marker.

12 > Change the **Warp Mode** to **Texture**, turn the **Flux** down to **0.00** and set the **Grain Size** to **25.00**. This gives us a good balance between a natural and a metallic sound for the vocal. Now drag the Warp Marker at the start of bar 2 over to the right – the further you drag it, the more stretched the vocal will become. Drag it to the start of bar 3. (Audio: **Stretched vocal.wav**)





13 > This gives us a classic stretched vocal effect, but let's put a new-school spin on it. It's not possible to automate the **Grain Size** or **Flux** in the arrangement, but we can automate them in the Sample Editor. Head over to the **Envelope** panel and change the **Control Chooser** to **Grain Size Modulation**.

14 > Copy the envelope curve shown above to decrease the **Grain Size** over the course of the sample. The result is a sinister effect, with the character of the timestretching changing as the sample plays back. Differently shaped curves will produce a variety of sounds, so don't be afraid to get experimental! (Audio: **Grain Size modulation.wav**)

POWER TIP

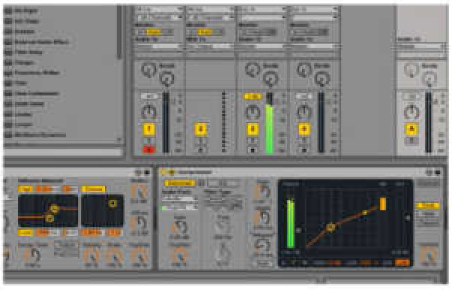
>Formant follies

If you enjoy creating unusual vocal effects, you should definitely spend some time checking out the **Complex Pro** Warp Mode. It's fundamentally similar to **Complex** mode but includes a parameter that adjusts how much formant compensation is applied when transposing the sample. It's intended to keep vocals sounding natural when transposed a few semitones, but at the maximum **Formant** value, shifting vocals down an octave or more creates some really quite unsettling, raspy, demonic voices. You can further affect the character of the sound with the **Envelope** fader – the lower this goes, the deeper into hell you will descend!

Advanced return processing

Return tracks are a convenient feature, to be sure. As we've seen in *Sends and returns* on p28, you can process multiple signals with the same effect using Return tracks to save on CPU, and even create feedback effect loops that are independent of the dry signal. It's this separation from the dry signal that makes Return tracks so powerful.

One of the most important things to remember when you're mixing a project is that the more elements you add to the mix, the messier the overall result will sound – especially if you're using reverbs and feedback delays, which can have long tails and an almost imperceptible but all-too-real impact on headroom. One solution to this is to simply use less elements and mix-clogging effects, but then



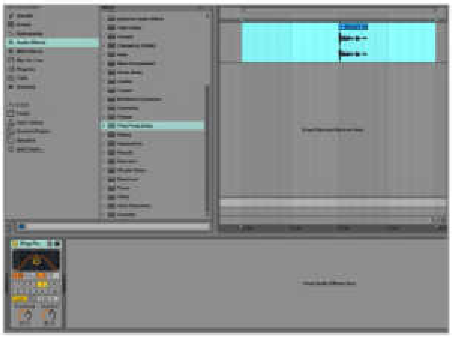
Sidechain compression is ideal for tidying up any reverb- or delay-related fuzziness on your return track

you run the risk of making your music sound plain and boring. Surely there must be some sort of middle ground...

Thankfully, there is! In *Sidechain routing* on p47, we used sidechain compression to sit an unruly kick drum and bassline combo together in the mix. The same principle can be applied to typical send effects, like reverbs and delays, which sound great but can make a mix less defined. This is especially problematic when working with vocals: it's likely you want the vocals in your track to be as intelligible as possible, and delays in particular can have an undesirable effect in this regard. By sidechaining the return channel with the input from the dry signal, we effectively take the messy delay out of the mix while the dry vocal is playing. When the vocal ends, the delay pops back into the mix: a quick and elegant solution that'll immediately make your mixes sound better.

> Step by step

5. Tidying up Return tracks with sidechain routing



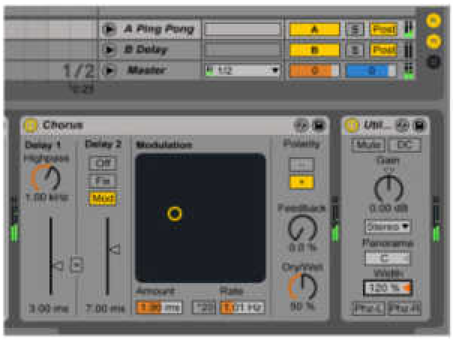
1 > Drag **Bangin' beats vocal.wav** onto an audio track. Click return track A to see the effects inserted onto it. There's a Reverb there by default, which we don't need, so click its Device Title Bar and press **Delete** to remove it. **Drag Ping Pong Delay** from the **Audio Effects** folder onto return track A.

2 > Turn the Ping Pong Delay's **Dry/Wet** knob up to **100%**, set **Beat Division** to **4** and **Bandwidth** to **9.00**, and turn the Return A send on the audio track up to **0dB**. On playback, the delay we've added makes the vocal significantly less intelligible. In the context of a mix, it's just going to sound like a mess, so we need to clean it up as much as possible. (Audio: **Messy vocal.wav**)

3 > Add Live's Compressor to the Return A track after the Ping Pong Delay. Click the **Sidechain Toggle** arrow at the top left-hand corner of its interface to bring up the Sidechain section, and click the **Sidechain** button located within to activate the effect's sidechain input.

4 > Click the **External Source** drop-down and select the vocal's audio track from the list. Bring the **Threshold** down to **-50dB**. This attenuates the delay when the dry vocal is playing, making it much easier to make out; but the release stage, where the delay comes back to full volume, is too obvious and sounds unnatural. (Audio: **Sidechained return.wav**)

5 > Turn the **Release** down to **10.0ms** and have another listen. Much better! We can improve things further by taking out some of the return track's low end with an EQ. Drag EQ Eight onto the return after the compressor, and click band 1's **Filter Selector** to select it. Change the band's filter mode to **Low Cut**, and its **Freq** to **1kHz**. (Audio: **Clean return.wav**)



6 > Another technique for getting the delay out of the way of the dry vocal is to move it out to the sides of the signal. A straightforward way to do this is to add a Chorus to the end of the effect chain. To make the effect more extreme, add a Utility after the Chorus and set its **Width** higher than **100%**; this will attenuate the mid signal. (Audio: **Wide send.wav**)

Additive synthesis with Operator

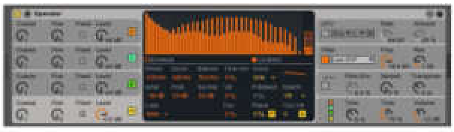
With its minimalist interface, Operator doesn't look like it's got much going on under the hood, but that couldn't be further from the truth. We've already talked about its impressive capabilities as a frequency modulation (FM) synth – see *Getting to grips with Operator* on p38 – but Operator also boasts additive synthesis functionality.

Much like FM synthesis, additive synthesis typically involves starting with a simple sine wave oscillator and adding 'partials' (additional sine oscillators) at different frequencies and amplitude levels. Many additive synths keep things simple, offering partials for a finite set of harmonics, and this is the system used by Operator, which allows partials on the first 16, 32 or 64 harmonics.

Partials are represented as a series of vertical bars, which can be dragged up and down to adjust their amplitude. Unfortunately, Operator doesn't offer a way to adjust the level of the partials in real time, so you're limited to creating static waveforms that can then be used in conjunction with the instrument's FM synthesis parameters, as an alternative to the usual sine and other shapes.

That may sound a little restrictive, but surprisingly, it actually enhances Operator's FM synthesis quite significantly, enabling all kinds of harmonic interest to be added to an oscillator before animating it by modulating it with another oscillator that varies in amplitude over time. The result? You get sounds that are much more complex than you might expect to hear from an FM synth – ideal for producing harmonically rich bass and lead sounds that can really stand out in a mix.

In the walkthrough below, we'll show you how to make your own additive oscillator waveform in Operator and bring it to life with a little frequency modulation magic, culminating in a rich timbre that can serve as the basis for a variety of patches.

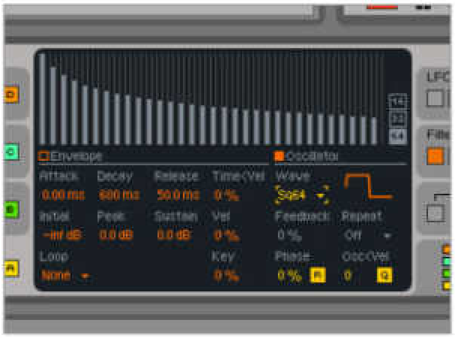


Harmonically complex waveforms are easy to generate using Operator's additive capabilities

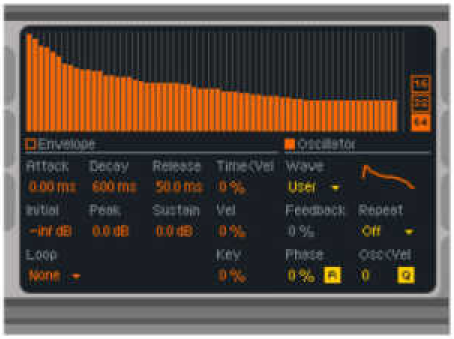
> Step by step

6. Creating a waveshape with Operator's additive synthesis

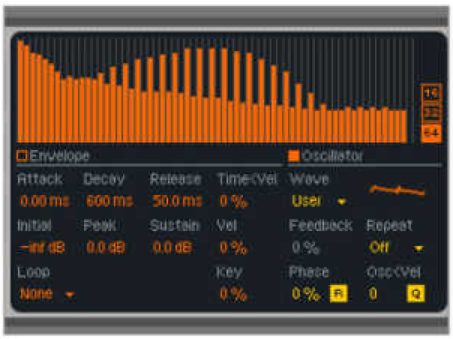
1 > Drag Operator onto a MIDI track and click the **Oscillator** button in the central panel. This brings up the oscillator's spectral Waveform Editor, which represents the waveform as a series of partials. Currently, the waveform is set to a sine wave shape, so you'll see that there's just a single partial at the fundamental frequency.



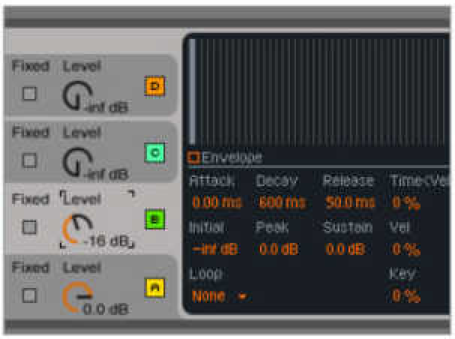
2 > We can see what more complex waveshapes look like by loading a different waveform into the oscillator. Click the **Wave** drop-down and select **Square 64** from the list of available shapes. As the Waveform Editor shows, this shape has partials on odd harmonics that decrease in amplitude the higher up the harmonic series they go.



3 > Editing partials is easy: simply drag them up for down. By dragging over the spectrum to fill in the gaps between the odd harmonics, we get something like a sawtooth waveform. The timbre will depend on the shape of the curve you've drawn. As you can see, making custom waveshapes is easy with Operator's additive synthesis capabilities. (Audio: **Drawn saw.wav**)



4 > Let's make something a little less virtual analogue-sounding. Right-click the Waveform Editor and select **Even**, which locks the odd harmonics so that we can only edit the even ones. Drag over the spectrum to boost the even harmonics in a curve, as shown above. This gives us a really rough and ready sound that will cut through a mix like nobody's business! (Audio: **Even harmonics.wav**)



5 > So, it's possible to make really potent waveforms with the Waveform Editor, but the problem is, they're static and thus don't sound terribly interesting. We can remedy this by combining additive synthesis with Operator's true raison d'être: FM synthesis. Turn Osc B's **Level** up as you play the patch back and you'll hear it really come to life.

6 > Set Osc B's **Level** to **-24dB**, then click the **Envelope** button in the central panel. Drag the **Attack** time up to **1.88s** to get a sweep with complex harmonics that can serve as the basis of a bass, lead or FX patch. Turn Osc B's **Coarse** level up to **6** to make it extra crunchy. (Audio: **Crunchy sweep.wav**)

Using Vocoder as an enhancer

Vocoders are great fun to play with and have provided robotic voice effects for the likes of ELO, Herbie Hancock, Beastie Boys and Daft Punk. But while synthetic-sounding vocals are what the vocoder is most famous for, they're by no means the only thing it's capable of. Live's Vocoder plugin, in particular, is an extremely flexible effect with plenty of uses, including the ability to make synthesised drum hits sound more realistic! This might seem crazy, but once you understand how vocoders work, the notion soon makes sense.

Vocoders impart the tonal characteristics of one sound (known as the 'carrier', often a synth) onto the articulation of another (the 'modulator', typically a vocal). The process works by splitting the modulator signal into a set of frequency ranges ('bands'), each of which is band-pass filtered and run through an envelope follower that detects the level of the band. The carrier is run through the same number of band-pass filters, and the envelope follower of each modulator band controls the amplitude level of the corresponding carrier band's output. This results in a sound that has the timbre of the carrier and the multiband dynamic response of the modulator.

A dedicated exciter is a great option for enhancing your drum sounds, but before you invest in one, you might want to try this clever technique for achieving the same sort of effect using Live's Vocoder...

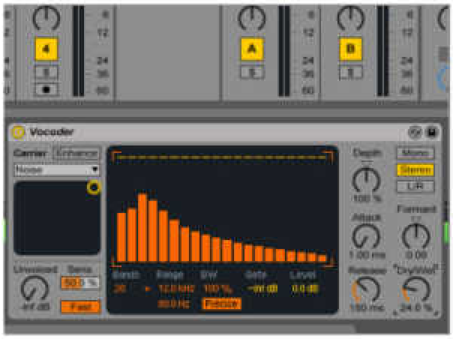
To use a vocoder to enhance a drum sound, white noise is employed as the carrier. White noise exhibits equal energy throughout its frequency range, so when it's used as a carrier, it mirrors the sound of the modulator. The less bands are used, the more noisy this mirrored sound becomes, adding a tonally bright 'aura' that reflects the timbre and movement of the synthetic drum and thus gives it a more harmonically complex, natural sound.

> Step by step

7. Enhancing a synthesised kick drum with Vocoder

1 > Before we can get busy with Vocoder, we need something to enhance, so let's make a kick drum sound with Operator. Drag Operator onto a MIDI track in Live's Session view. Click the **Pitch Envelope On/Off** button, and turn the **Pitch Envelope Amount** (**Pitch Env**) up to **100%.** Click Osc A, set its **Sustain** to **-inf dB**, and turn its **Release** up to **100ms**. This gives us a simple synthetic kick drum sound. (Audio: **Synth kick.wav**)

2 > It's tricky to make synthesised kicks sound like the real thing, but we can get a much more realistic result by using Live's Vocoder effect. Add Vocoder to the Operator track. If you play the Operator patch now, you'll hear something that sounds like howling wind in the higher octaves. This is because the Vocoder is set up to use its built-in noise carrier by default. (Audio: **Windy kick.wav**)



3 > We can use the noise carrier to give the kick a more natural-sounding, acoustic quality, but this requires a fair bit of subtlety. Turn the **Dry/Wet** knob down to **0.00%** so we can dial in just a little bit of noise: **24%** gives us an effect that feels right, without being too noticeable. (Audio: **Subtle noise.wav**)

4 > Adjust the timbre of the added noise by turning up the **Formant** knob a little – we've gone for **3.00**. The noise's tail is defined by the **Release** time – turn it up slightly to **175ms**. The effect can be made to sound more hi-fi or lo-fi by raising or lowering the number of **Bands**. Set it to **40** for maximum fidelity. (Audio: **Enhanced kick.wav**)

5 > It's hard to discern whether the subtle processing we're using is enhancing or weakening the sound, because it's now peaking at a lower level than it was prior to processing. To remedy this, add a Utility effect after the Vocoder, select them both, and press **Ctrl/Cmd+G** to group them together in an Audio Effect Rack. We're going to use Utility to make sure that the processed and unprocessed signals peak at the same level.

6 > Click the Operator track's **Peak Level** meter to reset it to **-inf dB**, and turn off the Audio Effect Rack. Compare the **Peak Level** with the Effect Rack turned off to the **Peak Level** with it turned on, and use the Utility's **Gain** to adjust the level of the signal when the effect is on – this will allow you to make a better-informed decision as to whether you've actually improved the sound or not. (Audio: **Level-matched enhanced kick.wav**)

Drum Rack choke groups and sends

Way back in *Sequencing your first beat with a MIDI track* on p11, we saw how to make a simple beat using a Drum Rack preset, and in *Audio slicing and creating custom Library content* on p19, we took things further, showing you how to use a Drum Rack to create your own Slice to New MIDI Track preset. But there are aspects of the Drum Rack that we haven't examined yet – its architecture is surprisingly deep!

Like Live's other Racks, the Drum Rack has the ability to host multiple chains. In the Drum Rack's case, each chain is triggered by a specific MIDI note, with multiple sounds playable at the same time by default. However, there are certain cases when you won't actually want two or more particular sounds to play together, the classic example being hi-hats. In a real-life drum kit, the hi-hat can either be open or closed – it can't be in both states at the same time. So, playing a closed hi-hat sample over the top of an open hi-hat sample results in a wholly inauthentic sound.

Unlike Schrödinger's cat, a hi-hat can only be in one state at a time. Avoid audio quantum confusion using the Drum Rack's choke feature!

This is where choke groups come in handy. Choke groups give you a way to tell a Drum Rack that only one sound in a given user-defined set can play at a time. So, if you start playing a closed hi-hat while an open hi-hat in the same choke group is still sounding, the open hi-hat will immediately stop.

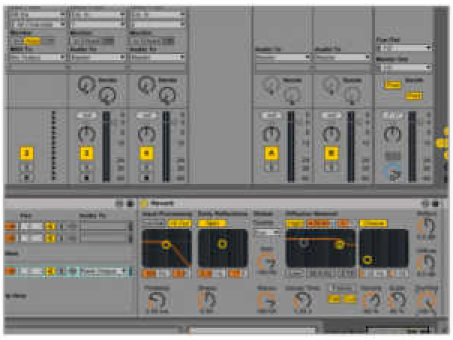
Another very cool feature of the Drum Rack is its integrated return channels. Of course, Live has its own return channels, but by creating returns within the patch itself, they can be saved as part of a preset, which can be a real time-saver if you use that patch regularly. What's more, these returns can be routed to Live's main returns, making it possible to apply the latter to individual drum sounds as well as the instrument's main out.

> Step by step 8. Advanced Drum Rack programming

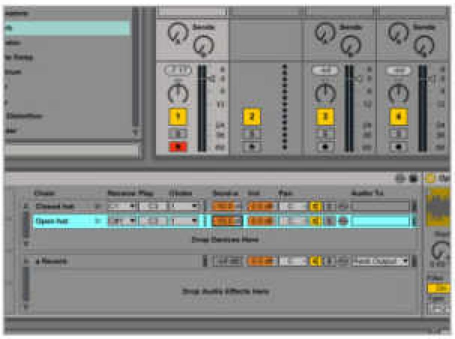
1 > You can find the Drum Rack in both the **Drums** and **Instrument** categories. Drag either version onto a MIDI track, then drag **Closed hat.wav** onto the C1 pad and **Open hat.wav** onto the C♯1 pad. Live will automatically create a Simpler instrument on each pad to play the samples back.

2 > Make a new MIDI clip and copy the sequence shown above. This sounds OK as it is, but choking the hi-hats will make it tighter and more authentic. Click the Drum Rack's **Show/Hide Chain List** button, then click the **Show/Hide Input/ Output Section**. (Audio: **Loose hats.wav**)

3 > Each chain has a **Choke Group** assignment, and by default these are all set to **None**. Set both chains' assignments to **1**. Now both hi-hats are in the same group, which means that as soon as one plays, the other stops dead. The upshot of this is that the open hi-hat's tail now sounds a good deal less messy. (Audio: **Tight hats.wav**)



4 > Click the **Show/Hide Return Chains** button. It's not possible to see the send routing parameters until we've created a return chain, so let's do that now. Drag **Reverb** from the **Audio Effects** folder onto the **Return Chains** panel (**Drop Audio Effects Here**), and turn its **Dry/Wet** knob up to **100%**.



5 > This creates a chain labelled 'a Reverb', and makes the **Show/Hide Sends** button above the **Show/Hide Return Chains** button available. Click it and a **Send-a** fader appears to the left of each chain's **Volume** fader. Turn both chains' **Send-a** faders up to **-10dB** to apply the new effect. (Audio: **Reverb send.wav**)

6 > You can also route individual chains to Live's main returns. To do this, right-click the **Return Chain** panel and select **Create Return Chain**. This creates an empty return chain. Change its **Audio To** routing to **B-Delay | Track In**, and you can use the newly created **Send-b** fader to apply the effect to a specific chain.

White noise-based sidechain effects

Sidechain compression is so ubiquitous that you'd be forgiven for assuming it's the be-all and end-all of sidechain-based processing. In fact, there are numerous other ways in which sidechaining can be used to enhance your mixes and for creative effect.

We've already encountered white noise in *Using Vocoder as an enhancer* on p142. In fact, Vocoder has its own built-in white noise generator that can be used instead of an external source, which should give you some idea of how important it is! But what is white noise, and why is it such as powerful tool?

In signal processing, 'noise' is a random signal, and its spectral density (ie, distribution of energy across the frequency spectrum) is defined by its colour. White noise has constant

White noise sounds like a detuned FM radio on its own, but in the right context it can be the icing on the cake!

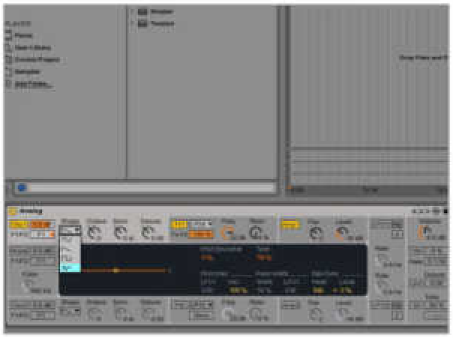
density across the entire spectrum, so it's as packed with information as an audio signal can possibly be. Sustained white noise doesn't actually sound very pleasant – it's just static, like an untuned FM radio – but when applied judiciously as part of a mix of signals, it can be extremely effective.

A common way to use white noise is to gradually open a low-pass filter over it, creating a 'riser' effect. Because these effects generally aren't mixed too loudly, and the high frequencies aren't present throughout the whole sound, it's not as fatiguing to listening to as loud, unfiltered white noise, but it generates a sense of fullness and excitement as the mix is filled up with energy.

In this walkthrough, we'll use the mix-filling energy of white noise to enhance a snare drum with a sidechained gate, and make a white noise riser where the filter's envelope follower is triggered from a sidechain input.

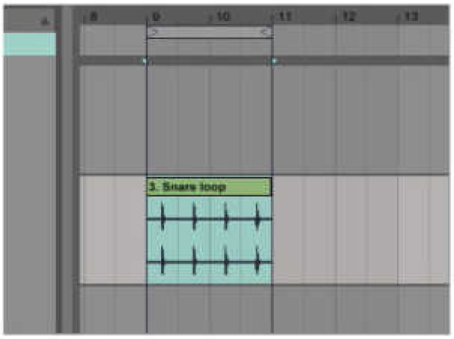
> Step by step

9. White noise effects with Analog and Gate

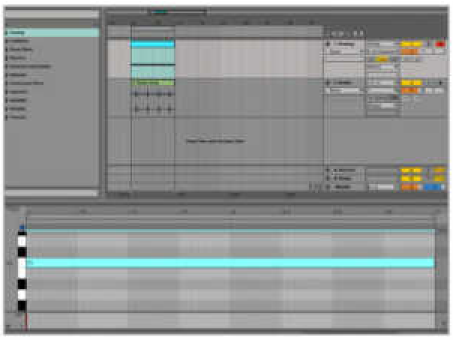


1 > Most synths can generate noise, so we'll keep things simple and use Analog. Drop it onto a MIDI track and deactivate Oscillator 2 by clicking the **Osc2** button. Click Oscillator 1's **Shape** drop-down, and select the **White Noise** waveform. This is the sound we want, but its amplitude envelope is a little lazy.

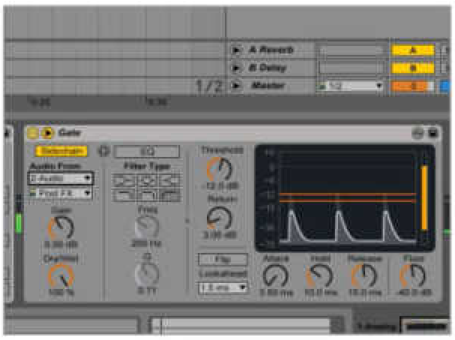
2 > Click the **Amp1** panel to bring up the amplitude envelope in the central panel. Turn the envelope's **Sustain** up to **1.00** and the **Release** down to **5ms**. This gives us a tight, loud white noise patch that we can use as a basis for a variety of effects. (Audio: **White noise.wav**)



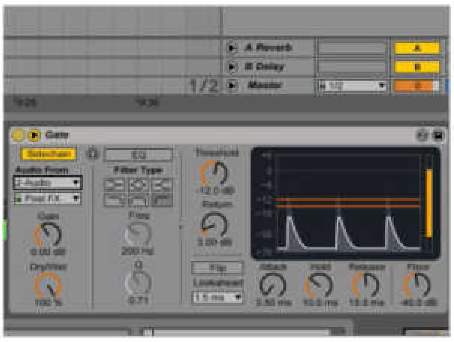
3 > Let's see how we can use a burst of white noise to enhance a snare drum. Drag **Snare loop.wav** onto an audio track. The snare has a bit of character and some weight in the low end, but it could do with a bit of beefing up in the highs.



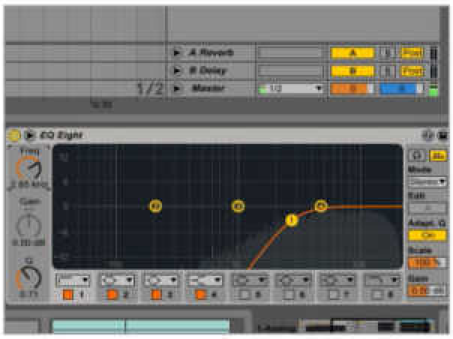
4 > Create a new MIDI clip of the same length as the snare loop and sequence a note playing throughout both bars. Because we're dealing with pure white noise, it doesn't matter what the note is. Naturally, having loud, unfiltered white noise playing alongside the snare sounds horrible, but we're working with very raw material at this point.

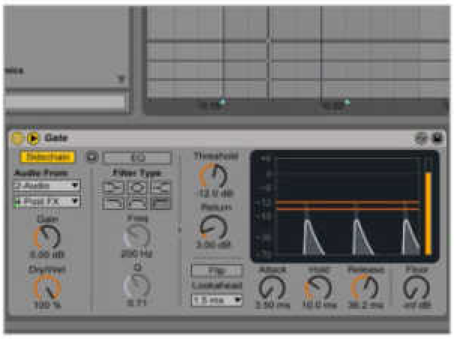


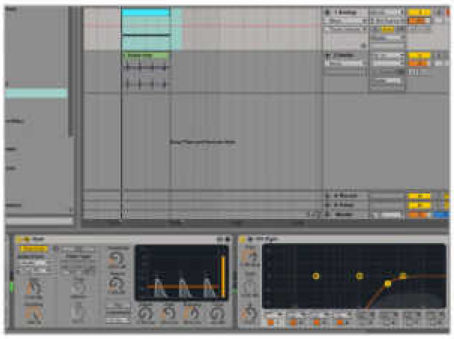
5 > Add Live's Gate effect to the Analog track, and click its **Sidechain Toggle** to bring up the **Sidechain** panel. Click the **Sidechain** button to activate the external sidechain input, and set the **Audio From** menu to the snare track. With the snare track feeding the Gate's envelope follower, it's controlling the level of the noise. (Audio: **Gated noise.wav**)



6 > Solo the Analog track and have a listen. You'll notice that although the noise peaks in volume when the snare hits, it's actually audible for the whole clip. This is because the Gate's **Floor** is set to **-40dB** by default. To silence the signal when the gate is closed, turn this down to **-inf dB**. (Audio: **Lower Floor.wav**)







7 > Unsolo the Analog track and take a listen to how both tracks sound at once. They sit together quite well, but we can make the sound a bit cleaner by adding an EQ Eight to the Analog track after the Gate, setting the first band to a low-cut filter and positioning it at **2.85kHz**. (Audio: **Clean snare**)

8 > We can further customise the white noise layer using the Gate's parameters. For example, try turning the **Release** up to **36.2ms** to give it a bit of a tail – like a subtle room reverb, but really clean and tight. (Audio: **Room effect.wav**)

9 > Control the intensity of the effect by adjusting the Gate's **Threshold**. The lower it goes, the longer the Gate will open for, and the more powerful the white noise will sound. A setting of **-30dB** will give you a big, bold snare that really cuts through the mix. Turning the Analog track down to **-5dB** balances the two and sounds a little better. (Audio: **Big snare.wav**)

10 > Auto Filter can also be used to create some nifty sidechained noise effects. Delete the Gate and EQ Eight effects on the Analog channel, and drop an Auto Filter on there instead. Replace the **Snare loop.wav** sample on the audio track with **Snare roll.wav**.

11 > Click the Auto Filter's **Sidechain Toggle** button, activate its sidechain input, and set **Audio From** to the snare track. Turn the **Filter Cutoff** down to **26.0Hz** to close the filter and prevent the noise from coming through.

12 > The snare track is feeding the Auto Filter's envelope follower, but as the **Envelope** parameter is currently set to **0.00**, it's not having any affect on the filter cutoff frequency. Turn the **Envelope** up to **127** and set the **Gain** to **11.0dB**. Mute the snare track.

13 > Now, on playback you'll be able to hear the snare roll opening the filter. To make the effect increasingly intense, automate the **Filter Cutoff** to move from **26.0Hz** at the start of the bar to **117Hz** at the end. Turn the **Envelope** level down to **40** to prevent the **Filter Cutoff** from peaking too soon.

14 > You can customise the sound's timbre and movement by tweaking the filter settings. Set the filter mode to **Bandpass**, turn the **Resonance** up to **3.00** and set the envelope **Attack** and **Release** times to **0.10ms** and **140ms** respectively. This gives us a lively, well-defined sound. (Audio: **Filter flutter.wav**)

POWER TIP

>Multiple sidechain input

Live's effects' sidechain inputs are limited to a single channel, but there is a workaround enabling two or more channels to be used. Create a return track and set its **Output Type** to **Sends Only**, then increase the **Send** level from all the tracks you want to use to that return track. Set the return track as the sidechain's input channel and you're ready to go! If you don't want the trigger channels to be heard on the master, set them to **Send Only** as well, as muting them will silence their sends too.

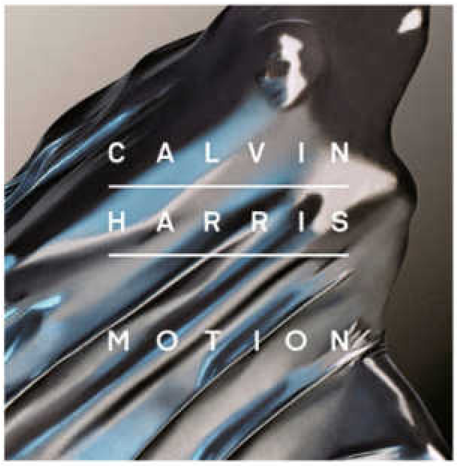
Setting up dedicated A/B tracks

Achieving the clarity, punch and loudness of a commercial master in your own productions is always a challenge, and it can be disheartening when you feel your work doesn't measure up to the competition.

An effective way to help you get closer to the kind of mixes and masters you want is to carry out a direct comparison within the project itself. Being able to hear the sonic characteristics of a production you consider to be exemplary can help you to focus your mixing decisions, and being able to make adjustments to anything from individual track levels to master buss processing right away is much more efficient than waiting until after you've completed a project to hold it up to comparative scrutiny.

There are plugins available that can help with so-called 'A/B' referencing, including Sample Magic's Magic AB, but the most straightforward solution is to create your own referencing track routing. A reference track is just a track that's routed directly to your audio interface's outputs. Say you've got a mastering plugin like iZotope Ozone on your master buss, and you're boosting the level of the track with its Maximizer. Any reference tracks would be subject to the same processing if they were just placed on any old track in the project, but by routing the audio to an output directly rather than via the master, you can hear it as intended.

Some veteran music industry professionals would strongly advise against putting *any* processing (in particular plugins that affect dynamics, like limiters or clippers) on your master buss, the argument being that it should be mastered by a professional, who needs plenty of dynamic range to work with in order to get the best result possible. These days, though, it's common for musicians to apply their own mastering processes, so that their music doesn't sound too quiet when posted online or played out. Thankfully, it's easy to bounce two versions of a mix: one with master buss processing, and one without that peaks at, say, -6dB to avoid the possibility of clipping. If the project sounds good both with and without the master buss processing, you've got all your bases covered!



A/Bing your tracks against some decent references takes the trial and error out of getting a polished sound, but be sure to choose something with a similar sound to the track you're working on!

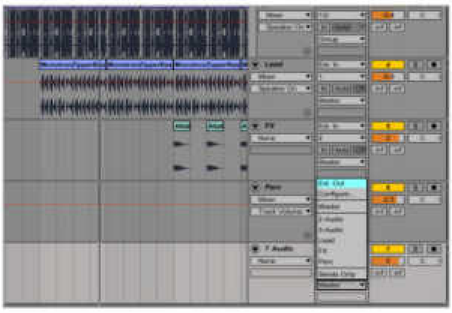
Character reference

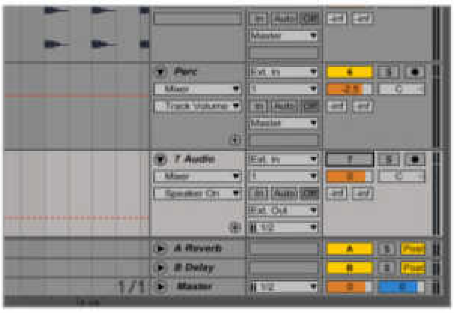
Unless you're going for a vintage-style or unorthodox sound, it makes sense to pick current tracks that are well produced, mixed and mastered as your reference material. It's also important to pick tracks that are similar in terms in the kind of sounds and mixing that they employ. Attempting to give your sultry electric piano-slathered slow jam the same kind of sonic characteristics as a sawtooth-packed, heavily sidechained EDM banger isn't necessarily going to make for the best end result. A similarly smooth groove would be a sensible choice.

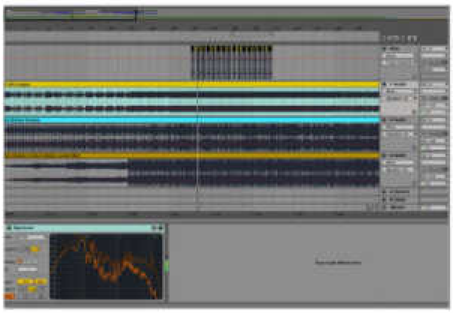
Referencing also works more effectively if you use several tracks that you feel accurately represent the sound you want to capture, rather than just one. This will give you an idea of the general ballpark you need to be aiming for, which is usually more effective than attempting to ape a particular track's sonic characteristics too closely.

If you find that you're unable to get your projects up to the same kind of level as your chosen commercial masters – despite limiting or clipping the master as hard as possible before it begins to have an adverse effect on the sound – you need to change something in your mix. You should be able to pick up clues as to the causes of any issues, as problem areas will being to distort first when you reduce the dynamic range on the master via clipping or limiting. So, if your kick and bassline are conspiring to create a horrible distorted mess, you know that you need to adjust their levels, or perform corrective operations such as equalising and/or sidechain compression to ensure that they don't clash.

> Step by step 10. Creating custom referencing tracks







1 > First, open a project that you're working on. If you're using any kind of processing on the master buss, this will affect any reference tracks routed to it. We want to avoid this, so we can get a totally accurate reference to the original file. Create an audio track and set its Output type to **Ext Out**.

2 > The Output Channel should be set to the same audio outputs as your master track. This will typically be 1/2, but you can check by looking at the **Master Out** settings on the Master track. Once the track is routed properly, click its **Track Activator** button to mute it.

3 > Duplicate this reference track a few times and drag some suitable pieces of music onto each copy, making sure to turn off warping for them. You can now A/B your mix with any of these tracks by soloing them and hearing how they compare. Put a Spectrum effect on your master and reference tracks and you'll be able to compare their frequency content visually, too. **cm**

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